IN THE UNITED STATES DISTRICT COURT

FOR THE EASTERN DISTRICT OF TEXAS

TEXARKANA DIVISION

ESN, L.L.C. . CIVIL ACTION NO. 5:08CV20

VS. . TEXARKANA, TEXAS

CISCO SYSTEMS, INC., ET AL. . JUNE 10, 2009

. 1:56 P.M.

TUTORIAL

BEFORE THE HONORABLE CHIEF JUDGE DAVID FOLSOM,

UNITED STATES DISTRICT JUDGE.

APPEARANCES:

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P R O C E E D I N G S

TEXARKANA, TEXAS

JUNE 10, 2009

(OPEN COURT)

THE COURT: PLEASE BE SEATED. GOOD AFTERNOON. WE ARE HERE FOR A TUTORIAL. I HAVE GIVEN EACH SIDE UP TO 45 MINUTES. ARE THE PARTIES READY TO GO FORWARD?

MR. MCANDREWS: WE ARE, YOUR HONOR.

MR. JONES: WE ARE, YOUR HONOR.

THE COURT: THEN PLAINTIFF MAY GO FORWARD.

MR. MCANDREWS: GOOD MORNING, YOUR HONOR. MY NAME IS PETER MCANDREWS. I REPRESENT ESN, THE PLAINTIFF IN THE CASE. I AM GOING TO PROVIDE WHAT I HOPE TO BE A BRIEF TUTORIAL ON THE SOLE PATENT-IN-SUIT, U.S. PATENT NUMBER 7,283,519, WHICH I WILL REFER TO AS THE '519 PATENT. I AM HOPING TO KEEP THIS BRIEF, AND IF YOU BELIEVE THAT FROM A PATENT ATTORNEY, THEN I HAVE A GOOD AUTOMOBILE DEALERSHIP I HAVE TO SELL YOU DOWN THE ROAD HERE. BUT I'LL KEEP IT AS BRIEF AS POSSIBLE. AND I THINK I CAN DO THAT BECAUSE THE UNDERLYING TECHNOLOGIES IN THIS CASE ARE REALLY NOT THAT COMPLEX TO UNDERSTAND AT THE LEVEL NECESSARY FOR OUR PURPOSES OVER THE NEXT TWO DAYS.

THE PARTIES ARE ALSO LARGELY IN AGREEMENT ON THE UNDERLYING TECHNOLOGIES. OF COURSE THERE ARE DISPUTES ABOUT THE PRECISE DEFINITIONS OF CERTAIN TERMS AND PHRASES, BUT THOSE WILL BE ADDRESSED IN THE CLAIM CONSTRUCTION HEARING

TOMORROW.

THE `519 PATENT IS ENTITLED: DISTRIBUTED EDGE SWITCHING SYSTEM FOR VOICE-OVER-PACKET MULTISERVICE NETWORK. IT WAS FILED IN APRIL OF 2002. IT ISSUED OCTOBER 16, 2007, AND THE SOLE INVENTOR IS GREGORY D. GIRARD OF MASSACHUSETTS.

THE TITLE OF THE PATENT IS ACTUALLY A GOOD PLACE TO START WITH THE TUTORIAL BECAUSE IT INTRODUCES SOME OF THE MOST BASIC CONCEPTS DEALT WITH BY THE `519 PATENT. ONE OF THE CONCEPTS IS FOUND IN THE TERM VOICE-OVER-PACKET MULTISERVICE NETWORK. THIS TERM REFERS TO THE FACT THAT THE `519 PATENT IS DIRECTED TO A SYSTEM FOR PROVIDING VOICE COMMUNICATIONS OVER A NETWORK THAT TRANSMITS VOICE DATA IN BUNDLES REFERRED TO AS PACKETS. THE TERM MULTISERVICE REFERS TO THE FACT THAT THE VOICE DATA PACKETS ARE TRANSMITTED OVER A NETWORK THAT IS ALSO USED TO COMMUNICATE OTHER TYPES OF DATA PACKETS SUCH AS EMAIL AND SO ON.

A UNIVERSALLY RECOGNIZED EXAMPLE OF SUCH A NETWORK IS THE INTERNET. VOICE COMMUNICATIONS OVER THE INTERNET IS COMMONLY REFERRED TO AS VOICE OVER INTERNET PROTOCOL, OR VOIP, V-O-I-P. ANOTHER BASIC CONCEPT INTRODUCED BY THREE WORDS OF THE TITLE, THE FIRST THREE WORDS, IS DISTRIBUTED EDGE SWITCHING. THIS PHRASE REFERS TO THE FACT THAT THE PATENT DISCLOSES A SYSTEM IN WHICH DEVICES ARE DISTRIBUTED ABOUT THE EDGE OF A NETWORK, FOR EXAMPLE, THE INTERNET. THESE DEVICES PROVIDE A SWITCHING FUNCTION. SWITCHING GENERALLY REFERS TO THE PROCESS OF

SETTING UP AND CONTROLLING A PHONE CALL BETWEEN TWO ENDPOINTS. THIS IS IN CONTRAST TO A SYSTEM SUCH AS THE PUBLIC TELEPHONE SYSTEM THAT I'LL DISCUSS IN A LITTLE MORE DETAIL IN A MINUTE AS BACKGROUND. THE PUBLIC TELEPHONE SYSTEM RELIES ON A SYSTEM OF CENTRALLY LOCATED SWITCHES TO PROVIDE TELEPHONE SERVICE.

NOW, IN THE BACKGROUND DISCUSSION OF THE PATENT, A FIGURE IS PROVIDED THAT DISCLOSES THE PUBLIC PHONE SYSTEM. IT'S CALLED THE PUBLIC SWITCHED TELEPHONE NETWORK OR PSTN. I'LL BE USING THAT ABBREVIATION, PSTN, THROUGHOUT THE NEXT SEVERAL MINUTES TO REFER TO THE TELEPHONE SYSTEM THAT MOST OF US HAVE KNOWN DURING MOST OF OUR LIVES, AND IT INCLUDES TO BE THE DOMINANT PHONE SYSTEM IN THE WORLD TODAY.

FIGURE 1 ILLUSTRATES SOME OF THE BASIC COMPONENTS OF THE PSTN. THERE ARE OF COURSE TELEPHONES AND THOSE ARE THE PHONES THAT YOU WOULD HAVE IN YOUR OFFICE, HERE IN THE COURTHOUSE, OR AT YOUR HOME. THE PHONES ARE CONNECTED TO SWITCHES. THESE SWITCHES AGAIN ARE THE SWITCHES THAT ARE GOING TO CONNECT ENDPOINTS, THE PHONES, TO EACH OTHER. THE SWITCHES ARE CONNECTED TO A SIGNALING NETWORK THAT ALLOWS THE SWITCHES TO COMMUNICATE INFORMATION TO EACH OTHER FOR THE PURPOSE OF ALLOWING ONE SWITCH TO CONNECT A CALL THROUGH TO ANOTHER SWITCH. AND FINALLY THERE IS A TRANSPORT NETWORK THROUGH WHICH A CIRCUIT IS MADE FOR COMMUNICATING VOICE SIGNALS BETWEEN A PHONE CONNECTED TO ONE SWITCH AND A PHONE CONNECTED TO ANOTHER SWITCH.

NOW, THE PSTN RELIES ON WHAT IS CALLED CIRCUIT SWITCHING. FOR PURPOSES OF THIS DISCUSSION IT IS MOST USEFUL TO UNDERSTAND CIRCUIT SWITCHING AS A CONTRAST WITH PACKET SWITCHING. NEWTON'S TELECOM DICTIONARY PROVIDES SUCH A DISCUSSION OF CIRCUIT SWITCHING IN ITS DEFINITION OF PACKET SWITCHING. AND IT SAYS: HERE IS ANOTHER WAY OF EXPLAINING PACKET SWITCHING: THERE ARE TWO BASIC WAYS OF MAKING A CALL. FIRST, THE ONE EVERYONE'S FAMILIAR WITH -- THE COMMON PHONE CALL. YOU DIAL. THE LOCAL SWITCH FINDS AN UNUSED PATH TO THE PERSON YOU CALLED AND JOINS YOU. WHILE YOU ARE SPEAKING, THE CIRCUIT IS 100 PERCENT ALL YOURS. IT'S DEDICATED TO THE CONVERSATION. THIS IS CALLED CIRCUIT SWITCHED. CIRCUIT SWITCHING IS THE WAY THE WORLDWIDE PHONE SYSTEM WORKS, ALSO CALLED THE PSTN. SO, CIRCUIT SWITCHING CREATES A CONNECTION OR CIRCUIT BETWEEN TWO PARTIES TO A TELEPHONE CALL WHERE THE CIRCUIT IS DEDICATED SOLELY TO THAT TELEPHONE CALL. IN OTHER WORDS, THE CIRCUIT IS NOT SHARED WITH ANY OTHER PHONE CALLS OR OTHER TYPES OF DATA.

THE PARTIES ARE IN GENERAL AGREEMENT ON THIS PART OF THE BACKGROUND AS IT RELATES TO CIRCUIT SWITCHING IN THE PSTN. CISCO'S EXPERT WITNESS, DR. BURGER, STATES: THE PSTN IS A CIRCUIT-SWITCHED NETWORK, WHICH MEANS THAT WHEN A TELEPHONE CALL IS CONNECTED, A CIRCUIT BETWEEN THE TWO TELEPHONES IS ESTABLISHED THAT IS DEDICATED SOLELY TO THE CALL. THAT CIRCUIT IS NOT SHARED WITH OTHER DATA.

NOW THE CIRCUIT SWITCHING IN A PSTN IS PERFORMED BY A SYSTEM OF CENTRALLY DEPLOYED SWITCHES. IN FACT, THE SWITCHES DEPICTED IN FIGURE 1 ARE REFERRED TO AS CENTRAL OFFICE SWITCHES. THEY ARE DEPLOYED IN WHAT IS COMMONLY REFERRED TO AS THE CENTRAL OFFICE. MOST TOWNS AND CITIES WILL HAVE ONE OR MORE CENTRAL OFFICES THAT HOUSE THESE SWITCHES. THESE FACILITIES ARE TYPICALLY OWNED AND OPERATED BY A LOCAL OR REGIONAL TELEPHONE COMPANY. IN FACT, OFTENTIMES THESE ARE THE VERY SAME BUILDINGS IN WHICH OPERATORS OF PHONE SYSTEMS PAST ACTUALLY RAN AROUND AND PULLED AND PUSHED PLUGS TO MAKE CONNECTIONS. SO IT'S USUALLY THE SAME CONCRETE BUNKER BUILDING, ONLY NOW THE WINDOWS ARE BLACKED OUT BECAUSE ALL YOU HAVE IN THERE IS A BUNCH OF COMPUTER HARDWARE.

IN THE PSTN THE ONLY EQUIPMENT THAT IS LOCATED ON A CUSTOMER PREMISE IS THE PHONE. THESE PHONES DO NOT HAVE TO INCLUDE ANY INTELLIGENCE BECAUSE THEY RELY ON A CENTRAL OFFICE SWITCH TO MAKE CONNECTIONS FOR THEM. THE PHONES ARE LITERALLY REFERRED TO AS PLAIN OLD TELEPHONE SERVICE TELEPHONES OR P-O-T-S, POTS PHONES. AND YOU WILL HEAR POTS PHONES REFERRED TO THROUGHOUT THE NEXT DAY OR TWO. THE TELEPHONES ARE CONNECTED OVER A WIRE, SUCH AS A COPPER TELEPHONE WIRE, TO THE REMOTE CENTRAL OFFICE SWITCH.

THE BASIC POTS TELEPHONE HAS A SPEAKER, A MICROPHONE, AND BUTTONS THAT GENERATE TONES WHEN THEY ARE PRESSED. THE TONES, SUCH AS THE TONES REPRESENTING A DIALED DIGIT OF A TELEPHONE

NUMBER OR THE POUND SIGN OR AN ASTERISK, ARE COMMUNICATED OVER THE PHONE WIRES TO THE CENTRAL OFFICE SWITCH WHERE THE SWITCH TRANSLATES THOSE TONES INTO A PHONE NUMBER TO ALLOW IT TO SET UP CALLS BETWEEN PHONES LOCATED ON DIFFERENT CUSTOMER PREMISES. THE OPERATION OF A POTS TELEPHONE IS SO BASIC THAT EVEN THE DIAL TONE IS PRODUCED BY THE CENTRAL OFFICE SWITCH.

USUALLY, PHONES IN THE SAME GEOGRAPHIC REGION, LIKE IN THE SAME TOWN, ARE CONNECTED TO THE SAME CENTRAL OFFICE SWITCH SO CALLS BETWEEN THOSE PHONES ARE SET UP AND CONTROLLED BY JUST A SINGLE SWITCH. AND IN THE FIGURE HERE, YOU CAN IMAGINE MULTIPLE PHONES ON THE RIGHT-HAND SIDE CONNECTED TO THE SAME SWITCH. HOWEVER, IF A CALL IS BEING MADE TO A PHONE CONNECTED TO A DIFFERENT CENTRAL OFFICE SWITCH, ADDITIONAL LAYERS OF SWITCHING ARE REQUIRED TO CONNECT ONE CENTRAL OFFICE SWITCH TO ANOTHER. AND THIS PASSAGE FROM THE `519 PATENT AT COLUMN 2, LINES 28 TO 37, EXPLAINS THE GENERAL OPERATION OF A CENTRAL OFFICE SWITCH. AND IN THE THIRD SENTENCE THERE IT SAYS: THE SIGNALING MODULE INTERFACES WITH THE SS7 -- SS#7 TRANSPORT NETWORK FOR THE PURPOSE OF SETTING UP A BEARER CHANNEL BETWEEN THE CALLING AND CALLED CENTRAL OFFICE SWITCHES. SO THE SS7 TRANSPORT NETWORK IS AN ADDITIONAL LAYER OF SIGNALING THAT ALLOWS SWITCHES TO MAKE CONNECTIONS BETWEEN SWITCHES FOR THE PURPOSE OF CONNECTING DISTANT PHONES TO EACH OTHER.

THE NEXT ITEM IN THE BACKGROUND THAT IS IMPORTANT TO UNDERSTAND IS PACKET SWITCHING. AND AS I DESCRIBED, PACKET

SWITCHING IS IN CONTRAST TO CIRCUIT SWITCHING. PACKET SWITCHING NETWORKS DEVELOPED IN PARALLEL TO THE PSTN AND THEY WERE MEANT FOR GENERAL DATA TRANSMISSION. THAT'S THE INTERNET WE KNOW TODAY. THE NEWTON'S TELECOM DICTIONARY, IN THE SAME DISCUSSION THAT I HAD EARLIER, THAT I PROVIDED EARLIER, ALTHOUGH IT WAS TRUNCATED TO LEAVE OUT THE PORTION DISCUSSING PACKET SWITCHING, STATES THAT: PACKET SWITCHING IS DIFFERENT. IN PACKET SWITCHING, THE CONVERSATION (WHICH MAY BE VOICE, VIDEO, IMAGES, DATA, ETCETERA) IS SLICED INTO SMALL PACKETS OF INFORMATION. EACH PACKET IS GIVEN A UNIQUE IDENTIFICATION AND EACH PACKET CARRIES ITS OWN DESTINATION ADDRESS -- I.E., WHERE IT'S GOING. EACH PACKET MAY GO BY A DIFFERENT ROUTE. THE PACKETS MAY ALSO ARRIVE IN A DIFFERENT ORDER THAN HOW THEY WERE SHIPPED. THE IDENTIFICATION AND SEQUENCING INFORMATION ON EACH PACKET LETS THE DATA BE REASSEMBLED IN THE PROPER SEQUENCE. PACKET SWITCHING IS THE WAY THE INTERNET WORKS.

NOW THE IDEA THAT DATA PACKETS MAY TRAVEL TO THEIR DESTINATION BY DIFFERENT ROUTES, ARRIVE OUT OF ORDER, AND EXPERIENCE DELAYS DUE TO OTHER DATA TRAFFIC IN THE SAME NETWORK LED SOME TO BELIEVE THAT PACKET SWITCHING WAS NOT A GOOD WAY TO COMMUNICATE LIVE OR REALTIME VOICE CONVERSATIONS SUCH AS WOULD TAKE PLACE DURING A TELEPHONE CALL. YOU CAN IMAGINE HOW FRUSTRATING IT MIGHT BE WHEN YOU SAY HELLO TO SOMEONE AND YOU DON'T GET THE HELLO COMING BACK TO YOU UNTIL LATER IN THE CALL AT SOME TIME.

HOWEVER, IMPROVEMENTS IN THE SPEED AND RELIABILITY OF THE INTERNET AND THE DEVELOPMENT OF EFFICIENT VOICE AND CODING TECHNIQUES EVENTUALLY LED TO THE INTRODUCTION OF VOICE OR VOICE OVER IP IN THE 1990S. THE NEWTON'S DICTIONARY COMMENTS ON THESE DEVELOPMENTS: RECENT DEVELOPMENTS OF CERTAIN SOFTWARE AND MAKING USE OF COMPLEX COMPRESSION ALGORITHMS, HOWEVER, HAS INTRODUCED PACKETIZED VOICE AND VIDEO TO THE CORPORATE INTRANETS AND THE INTERNET, WHICH WAS THE FIRST PACKET-SWITCHED NETWORK AND REMAINS BY FAR THE MOST HEAVILY USED TODAY.

AND AS A QUICK ASIDE, THIS MENTIONS COMPRESSION ALGORITHMS. WHAT THAT IS REFERRING TO IS VOICE ENCODING TECHNIQUES. AND WHAT A VOICE ENCODING SYSTEM DOES IS IT TURNS VOICE SIGNALS INTO DIGITAL DATA REPRESENTING, FOR EXAMPLE, THE FREQUENCY, VOLUME, AND OTHER PARAMETERS OF A VOICE AND THEN COMPRESSING THAT DATA SO THAT IT CAN BE EFFICIENTLY TRANSMITTED. THAT PARTICULAR IDEA IS NOT ALL THAT IMPORTANT FOR THE OVERALL CLAIM CONSTRUCTION ARGUMENT BUT IT'S A USEFUL PIECE OF INFORMATION TO HAVE.

NOW, NEWTON'S -- THE NEWTON'S DICTIONARY DEFINITION THAT I HAVE BEEN REFERENCING WAS PUBLISHED IN 2000 AND YET IT RECOGNIZES THAT VOIP IS ONLY A RECENT DEVELOPMENT EVEN AT THAT TIME. EVEN TODAY, THE MAJORITY OF PHONE SERVICE CONTINUES TO BE OVER THE CIRCUIT SWITCHED PSTN. THIS IS DUE IN PART TO THE FACT THAT VOIP WAS ORIGINALLY SOMEWHAT OF A NOVELTY AND

SYSTEMS HAD NOT YET BEEN DEVELOPED TO ALLOW VOIP TO MIMIC ALL OF THE FEATURES OF THE PSTN THAT THE PUBLIC HAD COME TO EXPECT.

IN THE MID TO LATE 1990S VARIOUS TELECOMMUNICATIONS STANDARD SETTING BODIES BEGAN WORKING ON SUCH SYSTEMS THAT WOULD ALLOW THE MIMICKING OF A PSTN. THESE SYSTEMS WERE GENERALLY REFERRED TO AS NEXT GENERATION NETWORK OR NGN. WHILE A HUGE NUMBER OF IDEAS WERE EXPLORED, FIGURE 2 OF THE `519 PATENT ILLUSTRATES SOME OF THE COMPONENTS OF SOME OF THE MOST PROMINENT IDEAS FOR THE NGN. THE NGN, AGAIN, STILL INCLUDED TELEPHONES AND, IN FACT, IT WAS ENVISIONED THAT THE PHONES WOULD STILL BE THE SAME POTS TELEPHONES THAT INDIVIDUALS HAD IN THEIR HOMES AND BUSINESSES. THE INDUSTRY RECOGNIZED THAT IF THE NGN WAS GOING TO BE WIDELY ACCEPTED, IT NEEDED TO ACCOMMODATE THE PHONES CUSTOMERS ALREADY HAD ON THEIR PREMISES.

TO ACCOMMODATE THESE PHONES, THOUGH, AN ADDITIONAL DEVICE CALLED A RESIDENTIAL OR MEDIA GATEWAY WAS DEPLOYED AT THE CUSTOMER PREMISES. THESE DEVICES INCLUDED THE VOICE ENCODERS AND DECODERS THAT I MENTIONED BEFORE REFERRED TO AS A KODAK CODER DECODER OF VOICES, OF A VOICE, AND MAKING THE VOICE PACKET SUITABLE FOR TRANSMISSION OVER THE INTERNET. HOWEVER, CONSISTENT WITH THE MODEL OF THE PSTN, THE RESIDENTIAL GATEWAY STILL RELIED ON CENTRALIZED NETWORK COMPONENTS TO SET UP AND CONTROL TELEPHONE CALLS.

AS THE '519 PATENT EXPLAINS IN ITS BACKGROUND SECTION, RESIDENTIAL GATEWAYS ARE UNINTELLIGENT IN THE SENSE THAT THEY REQUIRE THE MEDIA GATEWAY CONTROLLER, AND I'LL SHOW YOU WHERE THAT IS IN THE NETWORK IN JUST A MOMENT, TO MEDIATE ALL NETWORK SIGNALING FUNCTIONS ON THEIR BEHALF. THEY CANNOT DETERMINE THE BROADER NETWORK SIGNALING CONTEXT OF THE CALLING OPERATIONS IN WHICH THEY PARTICIPATE. THEY ARE INCAPABLE OF INDEPENDENTLY EXECUTING SERVICE LOGIC THAT INVOLVES NETWORK SIGNALING OPERATIONS, FOR EXAMPLE, CALL REDIRECTION, MULTIPOINT CALL CONTROL, CALL SUPERVISION, MULTIPLE LINE APPEARANCES, ETCETERA, WITHOUT CENTRALIZED PARTICIPATION BY SOMETHING CALLED A MEDIA GATEWAY CONTROLLER. THESE FACTORS IMPOSE SUBSTANTIAL CONSTRAINTS ON THE VARIETY OF NETWORK SERVICES THE NGN CAN DELIVER BECAUSE EACH NEW SERVICE MUST BE TIGHTLY INTEGRATED WITH THE MEDIA GATEWAY CONTROLLER IN ORDER TO PERFORM CALL CONTROL OPERATIONS.

AS ILLUSTRATED IN FIGURE 2, THESE MEDIA GATEWAY CONTROLLERS WERE DEPLOYED CENTRALLY. FOR EXAMPLE, THESE CONTROLLERS WOULD BE DEPLOYED IN THE SAME CENTRAL OFFICE THAT THE PSTN CENTRAL OFFICE SWITCH WAS DEPLOYED IN, OR AT YOUR INTERNET SERVICE PROVIDER. IN THE EARLY PART OF THE DECADE, THAT WAS LIKELY TO BE SOMEWHERE IN VIRGINIA AT AOL'S HEADQUARTERS. SO, THE CONTROL WAS VERY MUCH LEFT OUTSIDE THE CONTROL OF THOSE ON THE PREMISES. IN ANY EVENT, THE THEME OF CENTRALIZED CONTROL OF TELEPHONE CALLS WAS CARRIED FORWARD

FROM THE PSTN TO THE NGN.

NOW, A NUMBER OF DIFFERENT SIGNALING PROTOCOLS HAVE BEEN PROPOSED FOR THE NGN. A SIGNALING PROTOCOL ESSENTIALLY DEFINES THE VARIOUS ELEMENTS OF THE NETWORK AND THE LANGUAGE THAT THOSE ELEMENTS SPEAK TO EACH OTHER TO SET UP AND CONTROL PHONE CALLS. THE PRIMARY PROTOCOLS THAT HAD BEEN PROPOSED BY THE END OF THE `90S INCLUDED SOMETHING CALLED MGCP, WHICH IS MEDIA GATEWAY CONTROL PROTOCOL, H.323, AND SIP, SESSION INITIATION PROTOCOL, AMONG OTHERS. SIP, SESSION INITIATION PROTOCOL, I AM GOING TO SPEND A LITTLE MORE TIME IN A MOMENT ON BECAUSE THAT IS THE PRIMARY PROTOCOL RELIED ON BY THE INVENTION IN THE `519 PATENT.

WHILE EACH OF THESE PROPOSED PROTOCOLS HAD THEIR OWN PERCEIVED SET OF ADVANTAGES, THEY ALL SHARED A THEME THAT WAS COMMON TO THE PSTN. THE CRITICAL SYSTEMS NECESSARY TO SET UP AND CONTROL TELEPHONE CALLS, THE NETWORK INTELLIGENCE, REMAINED CENTRALLY DEPLOYED IN THE NETWORK. BECAUSE THE NGN WAS A NATURAL EVOLUTION FROM THE PSTN, IT WAS CONCEIVED AT THE OUTSET TO REALIZE SIMILAR ECONOMIES OF SCALE, LARGE SCALE UNIFORMITY OF SERVICE, AND A SIMILAR DEGREE OF CENTRALIZED MANAGEMENT CAPABILITY.

THESE THREE PROTOCOLS INCLUDED ROUGHLY ANALOGOUS GATEWAY DEVICES INTENDED FOR DEPLOYMENT AT A CUSTOMER'S PREMISE AND CENTRALIZED CONTROL ELEMENTS. MGCP, FOR EXAMPLE, HAD A MEDIA GATEWAY ON A CUSTOMER PREMISE AND IN THE CENTRAL OFFICE WAS

DEPLOYED A MEDIA GATEWAY CONTROLLER. IN H.323, THERE WAS A GATEWAY ON THE PREMISE AND A GATEKEEPER IN THE CENTRAL OFFICE. IN SIP WE HAVE AN ENTITY CALLED THE SIP USER AGENT THAT REPRESENTS THE ENDPOINT AND WOULD BE DEPLOYED IN ONE OF THE GATEWAYS ON THE PREMISE. AND IN THE CENTRAL OFFICE WE WOULD HAVE A SIP PROXY SERVER.

NOW, SIP IS AN ACRONYM FOR SESSION INITIATION PROTOCOL. AND AS THE NAME SUGGESTS, IT'S A PROTOCOL MEANT TO INITIATE SESSIONS. A SESSION, FOR EXAMPLE, IS A TELEPHONE CALL. SIP WAS INTRODUCED IN 1999 IN A DOCUMENT CALLED AN RFC. RFC LITERALLY MEANS REQUEST FOR COMMENT. IN RFC 2543 SESSION INITIATION PROTOCOL IS DESCRIBED AS AN APPLICATION LAYER CONTROL OR SIGNALING PROTOCOL FOR CREATING, MODIFYING, AND TERMINATING SESSIONS WITH ONE OR MORE PARTICIPANTS. THESE SESSIONS INCLUDE INTERNET MULTIMEDIA CONFERENCES, INTERNET TELEPHONE CALLS, AND MULTIMEDIA DISTRIBUTION.

THE FIRST DRAFT SIP STANDARD WAS PUBLISHED AS RFC 2543 IN MARCH OF 1999. AN UPDATED VERSION OF THAT DRAFT SIP STANDARD WAS PUBLISHED IN RFC 3261 IN JUNE OF 2002 SHORTLY AFTER THE `519 PATENT WAS FILED.

NOW AS I MENTIONED, THE NAME LITERALLY STANDS FOR REQUEST FOR COMMENTS. THE PURPOSE OF AN RFC IS TO SOLICIT COMMENTARY BY INDUSTRY EXPERTS SO THAT THE PROPOSED SIP PROTOCOL CAN BE IMPROVED AND ULTIMATELY BECOME A STANDARD. RFC 2543 STATES THAT THIS DOCUMENT SPECIFIES AN INTERNET STANDARDS TRACK

PROTOCOL, SO IT'S NOT A STANDARD YET. IT'S ON A STANDARD TRACK AND IT'S IN DRAFT FORM STILL. IT'S A STANDARDS TRACK PROTOCOL FOR THE INTERNET COMMUNITY AND REQUESTS DISCUSSION AND SUGGESTION FOR IMPROVEMENT. SO IT WAS ANTICIPATED, INDEED EXPECTED, THAT IMPLEMENTORS OF THE DRAFT STANDARD WOULD DEVIATE IN CERTAIN WAYS FROM THE DRAFT TO ALLOW EXPERIMENTATION WITH POSSIBLE IMPROVEMENTS.

NOW WE ARE GOING TO DISCUSS THESE DRAFT SIP STANDARDS IN A LITTLE MORE DETAIL TOMORROW, BUT I WANTED TO JUST SHOW YOU BRIEFLY GENERALLY WHAT A SIP STANDARD LOOKS LIKE. THIS IS LITERALLY THE FACE OF THE DOCUMENT. ON THE FACE IT'S REFERRED TO AS REQUEST FOR COMMENTS, NUMBER 2543. THIS IS ADDRESSING SIP, SESSION INITIATION PROTOCOL PUBLISHED IN MARCH OF `99. THE RFC THAT REPLACED IT IS RFC 3261 PUBLISHED IN JUNE OF 2002.

NOW I AM GOING TO COVER SOME OF THE FUNDAMENTAL BUILDING BLOCKS OF THE SIP PROTOCOL. THE FIRST ONE IS A SIP USER AGENT. A SIP USER AGENT, AS ITS NAME IMPLIES, IS THE ELEMENT OF A SIP NETWORK THAT IS INTENDED TO OPERATE ON BEHALF OF A USER, LIKE THE USER OF A TELEPHONE. IT'S THE ENDPOINT OF A SIP NETWORK THAT REPRESENTS NON-SIP THINGS AND HUMANS AND PHONES THAT EXIST OUTSIDE OF THE SIP NETWORK. BUT THE SIP NETWORK -- I AM SORRY. THE SIP USER AGENT IS ESSENTIALLY THE ENDPOINT OF THE SIP NETWORK.

NOW A SIP USER AGENT, THESE ARE SOME OF THE THINGS THAT

COME OUT OF THE DRAFT SIP STANDARD. A SIP USER AGENT, OR UA IN SOME CASES IT'S REFERRED TO, IS AN APPLICATION WHICH CONTAINS BOTH A USER AGENT CLIENT AND A USER AGENT SERVER. AND I AM GOING TO PROVIDE AN EXAMPLE IN A SECOND SO WE CAN UNDERSTAND A LITTLE BETTER WHAT THE DIFFERENCE BETWEEN THOSE TWO ELEMENTS IS.

WHEN THE SIP USER AGENT IS THE CALLING USER AGENT, IT IMPLEMENTS THE USER AGENT CLIENT WHICH IS A CLIENT APPLICATION THAT INITIATES THE SIP REQUEST SUCH AS AN INVITATION TO ANOTHER USER AGENT TO INITIATE A TELEPHONE SESSION. SO WHEN IT IS STARTING THE TELEPHONE CALL, IT ACTS AS A USER AGENT CLIENT. WHEN A SIP USER AGENT IS THE CALLED USER AGENT, IT IMPLEMENTS A USER AGENT SERVER WHICH IS A SERVER APPLICATION THAT CONTAINS -- I AM SORRY -- THAT CONTACTS THE USER WHEN A SIP REQUEST IS RECEIVED. IN OTHER WORDS, IT CONTACTS THE USER, FOR EXAMPLE, BY MAKING A TELEPHONE RING, AND THAT RETURNS A RESPONSE ON BEHALF OF THE USER. IN OTHER WORDS, AN INDICATION THAT, YES, NOW MY PHONE IS RINGING, YOU CAN GO AHEAD AND MAKE THAT RINGING SOUND IN THE PHONE SO YOU KNOW THAT THE PHONE IS RINGING AS OPPOSED TO BUSY, FOR EXAMPLE. THE RESPONSE THAT'S PROVIDED BY THE USER AGENT CLIENT WILL ACCEPT, REJECT, OR REDIRECT THE REQUEST.

AND HERE IS A BRIEF EXAMPLE, VERY MUCH SIMPLIFIED BUT IT SHOWS SOME OF THE BASIC SIGNALING THAT GOES ON TO SET UP A PHONE CALL. IN THIS EXAMPLE, WE HAVE SIP USER AGENT A ON THE

LEFT HAND AT THE TOP THERE. SIP USER AGENT A REPRESENTS AL'S ANALOG TELEPHONE. AND BETWEEN AL'S ANALOG TELEPHONE AND USER AGENT A, THERE IS NON-SIP DEVICE LEVEL SIGNALING, FOR EXAMPLE, THE THINGS WE MENTIONED EARLIER THAT A POTS PHONE CAN DO. IT CAN SEND OUT TONES THAT REPRESENT NUMBERS WHEN YOU PRESS THE BUTTONS OR IT CAN INDICATE THAT THE HANDSET HAS BEEN PICKED UP. ALL OF THAT INFORMATION IS WHAT IS REFERRED TO AS DEVICE LEVEL SIGNALING AND IT LETS THE USER AGENT KNOW WHAT IS GOING ON WITH THE POTS TELEPHONE.

SO, FOR EXAMPLE, WHEN AL PICKS UP HIS HANDSET AND DIALS BOB'S PHONE NUMBER, AND NOW BOB'S PHONE, I'VE ACTUALLY REPRESENTED BOB'S PHONE AS A NATIVE SIP PHONE. IN THIS CASE BOB'S PHONE ACTUALLY HAS A SIP USER AGENT BUILT INTO IT. SOMEBODY, YOU KNOW, PEOPLE HAVE MANUFACTURED PHONES TO INCLUDE THE SIP USER AGENT ENDPOINT RIGHT IN THE PHONE ITSELF. SO IN THIS CASE, AL PICKS UP HIS HANDSET AND DIALS BOB'S -- HE DIALS BOB'S PHONE NUMBER. SIP USER AGENT A ATTEMPTS TO INITIATE A SESSION BY SENDING AN INVITE MESSAGE TO USER AGENT B. AND THAT'S INDICATED AS NUMBER 1. AN INVITE IS SENT AND IT'S LITERALLY A MESSAGE THAT WILL INCLUDE THE TERM INVITE IN IT THAT IS SENT TO USER AGENT B.

IN THIS CASE, BECAUSE USER AGENT A IS THE CALLING USER AGENT, IT'S IMPLEMENTING THE USER AGENT CLIENT. AND BECAUSE USER AGENT B IS THE CALLED USER AGENT, IT IS IMPLEMENTING THE USER AGENT SERVER. SO USER AGENT B PROCESSES THE INVITE

MESSAGE, CAUSES BOB'S PHONE TO RING, AND THEN SENDS A RINGING MESSAGE TO USER AGENT A TO INDICATE THAT BOB'S TELEPHONE IS RINGING. SO THAT RINGING MESSAGE GOES BACK TO USER AGENT A AND USER AGENT A SENDS A SIGNAL TO AL'S ANALOG TELEPHONE AND SAYS, RING. IF BOB PICKS UP HIS HANDSET TO ANSWER, USER AGENT B WOULD SEND AN OKAY MESSAGE INDICATING THAT I HAVE PICKED UP THE PHONE, I AM HERE, I AM AVAILABLE TO ENGAGE IN A TELEPHONE SESSION. WHEN USER AGENT A RECEIVES THAT OKAY MESSAGE, IT WILL ACKNOWLEDGE RECEIPT OF THE OKAY MESSAGE AND SEND AN ACKNOWLEDGMENT OR AN ACK MESSAGE BACK TO USER AGENT B. THEREAFTER, A MEDIA SESSION, WHICH IS THE EXCHANGE OF THE DIGITIZED VOICE DATA, HAS BEEN SET UP AND IT WILL CONTINUE UNTIL ONE PARTY TERMINATES THE CALL BY HANGING UP.

NOW, IT'S NOT CRITICALLY IMPORTANT TO UNDERSTAND THIS POINT BUT OFTENTIMES THAT MEDIA SESSION, THE PASSAGE OF DIGITAL DATA ACROSS THAT MEDIA SESSION MAY IN SOME CASES FOLLOW A DIFFERENT PATH THAN THE SETUP MESSAGES. IT'S NOT IMPORTANT THAT THEY ARE PASSED THROUGH THE SAME ROUTERS AND SWITCHES. OFTENTIMES THAT MAY BE THE CASE. SOMETIMES IT'S NOT. THE FLEXIBILITY OF THE PROTOCOL ALLOWS YOU TO SET UP A PHONE CALL USING ONE DATA PATH BUT THEN COMMUNICATE THE DATA USING A DIFFERENT DATA PATH.

IN THIS CASE, AS I MENTIONED BEFORE, THE SIP ENDPOINT IS USER AGENT A. THE ENDPOINT IS NOT AL'S ANALOG TELEPHONE BECAUSE IT'S INCAPABLE OF COMMUNICATING SIP MESSAGES. AND AS

THE PATENT EXPLAINS, SIP USER AGENTS ARE CREATED TO OPERATE ON BEHALF OF TELEPHONE STATIONS, FOR EXAMPLE, NON-SIP TELEPHONES THAT ARE BY THEMSELVES INCAPABLE OF PERFORMING SIP NETWORK SIGNALING OPERATIONS. SO THE USER AGENT IS A SIP ENDPOINT.

THE '519 PATENT AND THE DRAFT SIP STANDARD REFER TO SIP USER AGENTS AS THE ENDPOINTS. RFC 3261 SAYS THAT THE SESSION INITIATION SIP PROTOCOL WORKS IN CONCERT WITH THESE PROTOCOLS AND THAT THESE, THE DISCUSSION HERE WAS A REFERENCE TO OTHER PROTOCOLS SUCH AS PROTOCOLS NECESSARY TO DIGITIZE AND COMPRESS VOICE DATA, BUT IN ANY EVENT, IT SAYS IT 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. THE CHARACTERIZATION OF A SESSION WOULD BE, FOR EXAMPLE, WE ARE GOING TO USE THIS TYPE OF VOICE COMPRESSION TECHNIQUE SO THAT EACH END OF THE CALL KNOWS ESSENTIALLY THE DIGITIZED VOICE LANGUAGE THAT EACH ENDPOINT WILL BE SPEAKING.

NOW ANOTHER FUNDAMENTAL BUILDING BLOCK OF THE SIP PROTOCOL IS A SIP PROXY SERVER, AND I MENTIONED THAT EARLIER AS THE CENTRALIZED ELEMENT. IN THE '519 PATENT YOU WILL FIND THE DEFINITION TAKEN FROM RFC 2543 EXACTLY. AND THE DEFINITION OF A SIP PROXY SERVER AS FOUND IN RFC 2543 IS AN INTERMEDIARY PROGRAM THAT ACTS AS BOTH A SERVER AND A CLIENT FOR THE PURPOSE OF MAKING REQUESTS ON BEHALF OF OTHER CLIENTS. REQUESTS ARE SERVICED INTERNALLY OR BY PASSING THEM ON

POSSIBLY AFTER TRANSLATION TO OTHER SERVERS. A PROXY INTERPRETS, AND, IF NECESSARY, REWRITES A REQUEST MESSAGE BEFORE FORWARDING IT.

NOW HERE IS A BRIEF ILLUSTRATION. MUCH LIKE THE EARLIER ILLUSTRATION, WE HAVE THE ENDPOINTS, USER AGENT A AND USER AGENT B, BUT IN THIS INSTANCE WE HAVE INSERTED A SIP PROXY SERVER AS AN INTERMEDIARY IN THE SIP COMMUNICATION. NOW THE SIP PROXY SERVER ACTS ESSENTIALLY AS A USER AGENT SERVER WHEN IT RECEIVES THE INVITE MESSAGE.

THE FIRST INVITE MESSAGE THERE LABELED NUMBER 1 WOULD BE ANALOGOUS TO THE INVITE MESSAGE WE SAW IN THE EARLIER ILLUSTRATION WHERE THERE WERE JUST TWO SIP ENDPOINTS. SO WHEN THE USER AGENT CLIENT PORTION OF USER AGENT A SENDS A MESSAGE, IT NEEDS A SERVER TO RECEIVE THAT. SO A USER AGENT SERVER PORTION OF THE SIP PROXY SERVER WILL RECEIVE THAT MESSAGE, BUT THEN THE SIP PROXY SERVER HAS TO TURN AROUND AND OUT THE OTHER SIDE ACT AS A SIP USER AGENT CLIENT. SO THE SIP PROXY SERVER, IN ITS ROLE AS ACTING AS AN INTERMEDIARY, ACTS AS BOTH A SERVER AND A CLIENT BUT IT'S ACTING AS A SERVER ON ONE SIDE OF THE CALL AND IT'S ACTING AS A CLIENT ON THE OTHER SIDE OF THE CALL. AND IT ACTS AS A CLIENT FOR PURPOSES OF PASSING ON THAT INVITE MESSAGE, WHICH IS THE MESSAGE NUMBER 3 THERE.

SO, IN THIS FASHION, SIP MESSAGES ARE RELAYED THROUGH THE INTERMEDIARY TO THE FAR END ENDPOINT, AND, BY THE WAY, THERE CAN BE MULTIPLE HOPS IN BETWEEN. THERE CAN BE MULTIPLE PROXY

SERVERS AS YOU GET CLOSER TO THE OTHER ENDPOINT. BUT THE POINT IS, THIS SIMPLE ILLUSTRATION SHOWS THAT THE SIP PROXY SERVER IS ACTING AS AN INTERMEDIARY.

ULTIMATELY WE SEE THAT THERE IS AN ACKNOWLEDGMENT MESSAGE IN THIS PARTICULAR ILLUSTRATION. IT'S SHOWING THE ACKNOWLEDGMENT MESSAGE SENT ALL THE WAY FROM ONE END TO THE OTHER. THAT WOULD BE IN A SITUATION WHERE THE ROUTING OF THE MEDIA PACKETS ARE GOING TO FOLLOW A DIFFERENT PATH THAN THE SETUP MESSAGES. SO YOU MIGHT LEAVE THE PROXY SERVER OUT OF THE MEDIA STREAM. IN SOME INSTANCES YOU MIGHT INCLUDE WHATEVER PIECE OF HARDWARE THE PROXY SERVER IS RUNNING ON, YOU MIGHT INCLUDE IT IN THE MEDIA STREAM. AND IN THAT CASE, YOU WOULDN'T SEE THAT ACKNOWLEDGMENT MESSAGE SKIPPING A STEP AND YOU WOULD SEE THE MEDIA SESSION PASSING FIRST THROUGH THE PROXY SERVER AND THEN TO THE FAR END.

NOW WITH THE TIME I HAVE LEFT HERE I AM GOING TO BRIEFLY GO THROUGH AN OVERVIEW OF THE TWO CLAIMS OF THE PATENT-IN-SUIT TO INTRODUCE YOU TO THE PROBLEMS THAT ARE SOLVED BY THE INVENTION. SO CLAIM 9 PROVIDES AN OVERVIEW. WE HAVE A NETWORK DEVICE, AND IN THE PATENT THAT NETWORK DEVICE IS REFERRED TO AS AN EDGE SWITCH. THE EDGE SWITCH HAS A BROADBAND NETWORK INTERFACE. THE BROADBAND NETWORK INTERFACE WILL BE ESSENTIALLY A MODEM THAT WILL CONNECT YOU TO THE BROADBAND ACCESS NETWORK. IN SOME HOMES YOU MAY HAVE A DSL MODEM. IN SOME HOMES YOU MAY HAVE A CABLE MODEM. BUT THE

IDEA IS IT IS THE CONNECTION BETWEEN YOU AND THE BROADBAND ACCESS NETWORK THAT WILL ULTIMATELY GIVE YOU ACCESS TO THE INTERNET.

THERE ARE A PLURALITY OF INTERFACES, INCLUDING A TELEPHONE LINE INTERFACE AND A COMPUTER DATA INTERFACE. THE TELEPHONE LINE INTERFACE WOULD BE, FOR EXAMPLE, THE INTERFACE THAT YOU PLUG YOUR POTS PHONE INTO SO THAT THE NETWORK DEVICE CAN PROVIDE AN INSTANCE OF A SIP USER AGENT TO REPRESENT THAT POTS PHONE. THERE IS A PROCESSOR AND THEN THERE IS A MACHINE-READABLE STORAGE MEDIUM THAT STORES PROCESSOR-EXECUTABLE INSTRUCTIONS, IN OTHER WORDS, SOFTWARE INSTRUCTIONS, TO PROVIDE SIP AGENTS THE INSTRUCTIONS CAUSING THE NETWORK DEVICE TO DO TWO THINGS. THE FIRST THING THAT IT DOES IS IT PROVIDES A SIP USER AGENT TO REPRESENT A NON-SIP TELEPHONE THAT USES THE TELEPHONE LINE INTERFACE. SO, FOR EXAMPLE, THERE IS A USER AGENT PROVIDED BY THIS NETWORK DEVICE FOR THE POTS PHONE THAT GETS PLUGGED INTO IT.

IN ADDITION, THIS NETWORK DEVICE, UNLIKE THE GATEWAYS OF THE PRIOR ART, INCLUDES A SIP PROXY SERVER THAT MEDIATES ALL SIP COMMUNICATIONS OVER THE BROADBAND NETWORK INTERFACE INVOLVING THE NON-SIP TELEPHONE. AND THAT SIP PROXY SERVER RUNS RIGHT ON THE NETWORK DEVICE.

THUS THE INVENTION OF CLAIM 9 PROVIDES BOTH A SIP USER AGENT AND A SIP PROXY SERVER IN THE SAME NETWORK DEVICE WHICH IS DEPLOYED ON THE SAME CUSTOMER PREMISE. THIS CAN BE SEEN IN

FIGURE 11 OF THE '519 PATENT WHERE EACH PREMISE HAS A NETWORK DEVICE WITH A SIP USER AGENT AND A SIP PROXY SERVER. THIS ALLOWS THE NETWORK DEVICES TO SET UP AND CONTROL TELEPHONE CALLS WITHOUT THE NEED FOR CENTRAL NETWORK ELEMENTS. THE MESH OF LIGHT GRAY DOTTED LINES REPRESENTS THE SIP SIGNALING PATHS THAT GO DIRECTLY FROM ONE PREMISE-BASED NETWORK DEVICE TO ANOTHER. THIS SYSTEM WAS CONSIDERED REVOLUTIONARY AT THE TIME BECAUSE IT WAS CONTRARY TO THE STANDARD MODEL OF CENTRALIZED CONTROL CARRIED FORWARD FROM THE PSTN. WITH THIS SYSTEM, CUSTOMERS AND BUSINESSES WOULD BE ABLE TO HAVE TELEPHONE SERVICE WITH ALL THE BENEFITS AND FEATURES OF THE PSTN WITHOUT CEDING CONTROL TO A TELEPHONE CARRIER AND WITHOUT HAVING TO PAY FOR ANYTHING OTHER THAN INTERNET ACCESS.

NOW WITH THE LEVEL OF AUTONOMY PROVIDED BY THE INVENTION OF CLAIM 9, THE INVENTOR ALSO RECOGNIZED THAT THERE WAS STILL A NEED FOR SOME LEVEL OF CENTRALIZED MONITORING OF THE PHONE CALLS BEING MADE. FOR EXAMPLE, IF A BUSINESS WANTED TO KEEP TRACK OF THE TIME, DATE, AND LENGTH OF PHONE CALLS FOR THE PURPOSE OF MONITORING ITS EMPLOYEES OR EVEN CHARGING CUSTOMERS, THERE WAS SOME LEVEL OF CENTRALIZED OVERSIGHT THAT MIGHT BE DESIRED. SO CLAIM 16 PROVIDES A METHOD OF COLLECTING CALL LOG DATA FROM THE VARIOUS NETWORK DEVICES. CLAIM 16 IS A METHOD. CLAIM 16 WILL INCORPORATE IN LARGE PART THE DEVICE THAT WAS INTRODUCED IN CLAIM 9. BUT IN ANY EVENT, THIS IS A METHOD CLAIM AND IT CALLS FOR LOCATING A SYSTEM MANAGEMENT

PLATFORM IN A SHARED PACKET NETWORK. THE SYSTEM MANAGEMENT PLATFORM COLLECTING CALL LOG DATA FROM A PLURALITY OF NETWORK DEVICES. SO THE SYSTEM MANAGEMENT PLATFORM WOULD BE CENTRALLY LOCATED, FOR EXAMPLE, IN A CENTRAL OFFICE. AND THEN IN ADDITION, A NUMBER OF NETWORK DEVICES WILL BE DISTRIBUTED ABOUT THAT NETWORK. AND FIGURE 4 SHOWS THREE INSTANCES OF A NETWORK DEVICE. THESE NETWORK DEVICES ARE CAPABLE OF ROUTING CALLS IN A PEER-TO-PEER FASHION, WHICH IS ANOTHER WAY OF SAYING THAT THEY CAN ROUTE CALLS FROM ONE PREMISE TO ANOTHER WITHOUT THE NEED FOR CENTRALIZED CONTROL. AND FINALLY, CLAIM 16 ADDS THE REQUIREMENT OF A PROXY SERVER IN EACH OF THESE NETWORK DEVICES. THAT'S ALL I HAVE, YOUR HONOR.

THE COURT: THANK YOU. WHY DON'T WE TAKE A FIVE MINUTE RECESS FOR THE DEFENDANT TO SET UP, AND THEN WE WILL HAVE THE TUTORIAL FROM CISCO. WE WILL BE IN RECESS FIVE MINUTES.

(RECESS AT 2:31 P.M., UNTIL 2:37 P.M., OPEN COURT)

THE COURT: PLEASE BE SEATED. READY TO GO FORWARD ON BEHALF OF CISCO.

MS. SHARPER: YES, I AM. GOOD AFTERNOON. MY NAME IS SAYURI SHARPER AND I REPRESENT THE DEFENDANTS, CISCO SYSTEMS AND CISCO-LINKSYS.

HERE IS THE OUTLINE OF WHAT I AM GOING TO COVER TODAY. WE ARE GOING TO START WITH THE BACKGROUND TECHNOLOGY, HOW THE VOICE NETWORK HAS EVOLVED FROM PSTN TO VOICE OVER IP. AND WE

ARE GOING TO FOCUS FOR VOICE OVER IP TWO ALTERNATIVE ARCHITECTURES. THERE IS A TELECOM-CENTRIC ARCHITECTURE AND THE INTERNET-CENTRIC ARCHITECTURE.

AFTER THE BACKGROUND SECTION, WE WILL FOCUS ON SIP. SIP IS SESSION INITIATION PROTOCOL. AND THERE ARE SIP BUILDING BLOCKS THAT ARE DEFINED BY THE SPECIFICATION TO EXPLAIN HOW YOU WOULD BUILD A VOICE OVER IP NETWORK USING SIP. WE WILL TAKE A LOOK AT THAT, TAKE A LOOK AT HOW SIP MESSAGES WORK, AND HOW THESE MESSAGES ARE USED BY THESE BUILDING BLOCKS TO INVOKE SIP OPERATIONS.

AND THEN WE WILL TAKE A LOOK AT THE '519 PATENT, REVIEW A LITTLE BIT OF THE PRIOR ART THAT IS MENTIONED IN THE PATENT AND FOCUS ON HOW SIP IS USED IN THE PATENT ITSELF.

SO THIS IS A PICTURE OF A TRADITIONAL PUBLIC SWITCH TELEPHONE NETWORK, VERY SIMILAR TO WHAT MR. MCANDREWS WENT THROUGH IN HIS TUTORIAL SESSION. THE TRADITIONAL PUBLIC SWITCH TELEPHONE NETWORK USED CIRCUIT-SWITCHED SYSTEMS. I WON'T GO OVER THAT AGAIN BECAUSE THAT WAS PRETTY WELL COVERED. THE ONE THING I WOULD LIKE TO POINT OUT IS THE CLASS 5 SWITCH OR THE SWITCH IN THE CENTRAL OFFICE. THE POINT HERE IS THAT WITHOUT A SWITCH IN THE NETWORK, A TELEPHONE IN A HOUSE CAN'T TALK TO ANOTHER TELEPHONE. IT'S REQUIRED TO MAKE A CALL CONNECTION BETWEEN THE TWO TELEPHONES FOR YOU TO HAVE A CONNECTION BETWEEN THEM.

SO PRIMARILY THE NETWORK THAT IS OUT IN THE PUBLIC WAS

THIS PSTN, BUT PSTN IS REALLY NOT SUITED TO TRANSPORTING DATA. IN THE 1990S INTERNET BECAME VERY POPULAR. AND IN ORDER TO TRANSPORT DATA, WHAT YOU NEED IS A PACKET-SWITCHED NETWORK. SO PEOPLE STARTED BUILDING PACKET-SWITCHED NETWORK INFRASTRUCTURE SO THAT THEY CAN ROUTE THIS DATA TRAFFIC. AGAIN, THE TERMS OF THE BASIC TECHNOLOGY OF PACKET-SWITCHED NETWORK THAT HAS BEEN COVERED BY MR. MCANDREWS, SO I WON'T GO OVER THAT HERE. SUFFICE IT TO SAY THAT A LOT OF INVESTMENT WAS PUT IN PLACE TO BUILD THIS SECOND NETWORK TO TRANSPORT DATA IN ADDITION TO VOICE.

SO SOME OF THESE SERVICE PROVIDERS WHO WERE OFFERING DATA SERVICE STARTED TO WONDER, HEY, CAN WE GET VOICE INTO OUR NETWORK ALSO FOR ADDITIONAL REVENUE? CAN WE SELL THIS NETWORK SERVICE FOR PEOPLE TO ROUTE THEIR VOICE DATA? AND THAT'S THE START OF VOICE OVER IP.

SO IN ORDER TO ACCOMMODATE THAT, WHAT YOU NEEDED TO DO IS TO CHANGE THIS CONTINUOUS STREAM OF VOICE TRAFFIC AND PUT THAT INTO INDIVIDUAL DATA PACKET CHUNKS. SO MAKE VOICE LOOK MORE LIKE DATA AND THEN YOU CAN ROUTE THAT IN THE INTERNET OR ANY PACKET-SWITCHED NETWORK JUST AS IF YOU ROUTE DATA. SO THAT WAY A PHONE THAT IS CONNECTED TO AN INTERNET CAN TALK TO ANOTHER PHONE CONNECTED TO THE INTERNET.

THAT SERVICE, HOWEVER, WASN'T TOO APPEALING BECAUSE A LOT OF PEOPLE WERE STILL SERVED OUT OF THE PUBLIC SWITCH TELEPHONE NETWORK. SO IF I CAN ONLY CALL MY NEIGHBOR WHO IS ON THE SAME

NETWORK AS ME, THERE IS ONLY SO MUCH VALUE. IN ORDER TO ACCOMMODATE MY ABILITY TO CALL ANYBODY, WHAT I NEEDED WAS A GATEWAY THAT COULD CROSS BETWEEN THESE DIFFERENT TYPE OF NETWORKS. SO THAT'S THE REASON OF HAVING THIS PSTN GATEWAY WHICH IS IN THE CENTER OF THE DIAGRAM. WHAT IT DOES, IT CONVERTS PACKET DATA -- PACKET VOICE USING VOICE OVER IP INTO THE FORMAT THAT'S COMPATIBLE WITH A STANDARD TELEPHONE NETWORK.

SO HERE IS A TIMELINE OF HOW VOICE OVER IP EVOLVED. THE FIRST COMMERCIALLY AVAILABLE PRODUCT THAT SUPPORTS VOICE OVER IP CALLED INTERNET PHONE WAS INTRODUCED TO THE MARKET IN FEBRUARY OF 1995. AND THIS IS A PRODUCT THAT WAS OFFERED BY VOCALTEC. SO USING THE INTERNET PHONE YOU CAN CALL PEOPLE THAT WAS ATTACHED TO INTERNET WHO ALSO HAVE THAT SOFTWARE. SO THIS WAS GREAT. IT ENABLED PEOPLE THAT WAS OFFERING INTERNET SERVICE TO PROVIDE VOICE CAPABILITY AS WELL.

THE VOCALTEC PRODUCT, HOWEVER, WAS PROPRIETARY. IT USES ITS OWN PROTOCOL WHICH MEANT THAT ALL THE EQUIPMENT HAD TO BE ACQUIRED FROM VOCALTEC. SO THAT LIMITED THE APPEAL OF THE PRODUCT BECAUSE IF YOU WANT TO DESIGN A LARGE NETWORK, WHAT YOU WANT IS TO BE ABLE TO BUY EQUIPMENT FROM MULTIPLE VENDORS.

SO STANDARDS SETTING ORGANIZATIONS STARTED WORKING ON VOICE OVER IP. THERE ARE TWO DIFFERENT STANDARD SETTING ORGANIZATIONS THAT WAS WORKING ON THIS. ONE IS ITU-T AND THE OTHER IS IETF. SO THE FIRST SET OF STANDARDS WAS PUBLISHED BY

ITU-T IN NOVEMBER OF 1996 AND THIS IS KNOWN AS THE H.323 RECOMMENDATIONS. THE IETF PUBLISHED A DIFFERENT PROTOCOL CALLED SIP AS RFC 2543 IN MARCH OF 1999. ITU-T PUBLISHED ANOTHER PROTOCOL KNOWN AS MEGACO, OR H.248 IN JUNE OF 2000.

SO BY THE END OF 2000, THERE WERE TWO COMPETING VOICE OVER IP ARCHITECTURES ON THE MARKET. ONE WAS BASED ON THE SET OF STANDARDS FROM ITU-T, SO THESE ARE THE H.323 AND THE H.248 OR MEGACO STANDARDS. AND THE OTHER ONE WAS THE INTERNET-CENTRIC SIP PROTOCOL.

SO WHAT ARE THESE STANDARDS ORGANIZATIONS? JUST TO GO OVER THEM BRIEFLY, ITU-T IS AN ORGANIZATION WITHIN THE INTERNATIONAL TELECOMMUNICATIONS UNION WHICH IS PART OF THE UNITED NATIONS. AND ITS PURPOSE IS TO COORDINATE STANDARDS FOR TELECOMMUNICATIONS. SO THE KEY HERE IS TO UNDERSTAND THAT THE HERITAGE OF ITU-T IS FROM THE TELECOMMUNICATIONS INDUSTRY. SO IT'S A MEMBERSHIP DRIVEN ORGANIZATION. IN ORDER FOR YOU TO CONTRIBUTE OR PUT -- HAVE COMMENTS ON THEM, YOU HAVE TO BE A MEMBER. AND THE MEMBERSHIP CONSISTS OF 191 COUNTRIES AND ABOUT 700 COMPANIES. SO AGAIN, THE FOCUS HERE IS TO MAKE SURE THAT THE TELECOMMUNICATIONS NETWORK AROUND THE WORLD INTEROPERATES.

ITU-T STANDARDS ARE KNOWN AS RECOMMENDATIONS. RECOMMENDATIONS ARE DEVELOPED IN STUDY GROUPS. THEY GO THROUGH A REVIEW PROCESS WHERE ANY MEMBER CAN SUBMIT THEIR COMMENTS. ONCE THOSE ARE RESOLVED, THEN IT'S PUBLISHED AS A

RECOMMENDATION. STANDARDS, HOWEVER, EVOLVED. THERE IS NO STANDARD THAT'S STABLE BECAUSE THINGS CHANGE. SO HOW ITU-T ACCOMMODATE THAT IS THAT THEY REVISE THE STANDARD WITH NEW VERSIONS. SO, FOR EXAMPLE, H.323 WE SAW THE FIRST VERSION OF THE SPECIFICATION WAS PUBLISHED IN 1996. THE LATEST VERSION, WHICH IS VERSION 6, WAS PUBLISHED IN 2006, SO THAT YOU CAN MODIFY THE STANDARD AS YOU LEARN NEW THINGS AND AS A REQUIREMENT CHANGES.

THIS IS IN CONTRAST TO IETF WHICH IS THE INTERNET ENGINEERING TASK FORCE. THE FOCUS OF THIS GROUP IS IN THE INTERNET AS OPPOSED TO TELECOMMUNICATIONS. AND RATHER THAN BEING A MEMBERSHIP DRIVEN ORGANIZATION, THIS IS AN OPEN ORGANIZATION WHICH MEANS THAT ANYBODY CAN ATTEND THEIR MEETINGS AND ANYBODY CAN MAKE COMMENTS. IT'S NOT RESTRICTED IN ANY WAY TO JUST THE MEMBERS. EVEN THOUGH IT SOUNDS LIKE THIS IS A MORE INFORMAL ORGANIZATION, IETF ACTUALLY FOLLOWS A VERY RIGOROUS STANDARDIZATION PROCESS.

SO ALL STANDARDS OR ACTUALLY ALL SPECIFICATIONS THAT ARE BEING PROPOSED AS STANDARD HAS TO BE SUBMITTED TO IETF FIRST AS AN INTERNET-DRAFT. THE INTERNET-DRAFT THEN IS -- BECOMES PUBLICLY AVAILABLE AND WILL BE OPEN FOR REVIEW AND COMMENTS BY ANY INTERESTED PARTY. IN ORDER TO RESOLVE DIFFERENCES OF OPINIONS ABOUT THE SPECIFICATION, THIS WORK IS HANDLED WITHIN A WORKING GROUP. THIS IS SIMILAR TO THE STUDY GROUP IN ITU-T. SO THE WORKING GROUP GETS TOGETHER, THEY REVISE THE

SPECIFICATION, AND KEEP ON WORKING AT IT UNTIL THERE IS CONSENSUS ON THE SPECIFICATION ITSELF. ONCE THERE IS CONSENSUS, THEN THE DRAFT SPECIFICATION IS SUBMITTED TO THE INTERNET ENGINEERING STEERING GROUP FOR APPROVAL.

THERE IS A SPECIFIC CRITERIA THAT NEEDS TO BE MET BEFORE ANY DRAFT SPECIFICATION CAN BE APPROVED AS A STANDARD. THE FIRST ONE IS THAT THE SPECIFICATION HAS TO BE WELL UNDERSTOOD. THAT MEANS THE SPECIFICATION HAS TO BE CLEAR TO SOMEONE WHO READS IT. IT HAS TO HAVE RECEIVED SIGNIFICANT COMMUNITY REVIEW. SO WITHOUT THE REVIEW PROCESS IT WILL NOT BE APPROVED. AND IT HAS TO ENJOY ENOUGH COMMUNITY INTEREST TO BE CONSIDERED VALUABLE. THESE ARE PROTOCOLS THAT WILL BE USED ON THE INTERNET. AND THEY JUST DON'T APPROVE ANYTHING YOU SUBMIT. IT HAS TO BE OF VALUE TO THE OPERATION OF THE INTERNET. SO ONCE IT'S SUBMITTED AND APPROVED, THEN THE SPECIFICATION IS PUBLISHED WITH AN OFFICIAL RFC NUMBER.

UNLIKE THE ITU-T, THE STANDARDS DOCUMENT WITHIN IETF DOES NOT HAVE REVISION NUMBERS. SO IF YOU HAVE AN RFC PUBLISHED, THAT DOCUMENT NEVER CHANGES. HOWEVER, AS WE SAID, STANDARDS EVOLVE SO THEY HAVE TO HAVE A MECHANISM TO ACCOMMODATE THAT. SO WITHIN IETF WHEN THERE IS A REVISION TO A STANDARD, IT'S PUBLISHED WITH A NEW RFC NUMBER. SO WHAT DOES RFC MEAN? I THINK WE'VE HEARD COMMENTS THAT REQUEST FOR COMMENTS ARE NOT FINAL DOCUMENTS BECAUSE IT STANDS FOR REQUEST FOR COMMENTS, SO OBVIOUSLY IT'S NOT FINAL. BUT THAT REALLY IS NOT HOW THE

TERMINOLOGY IS USED WITHIN IETF. WITHIN IETF THE NOMENCLATURE FOR REQUEST FOR COMMENTS IS REALLY AN OFFICIAL IETF PUBLICATION.

SO ANYTHING THAT IETF PUBLISHES HAS AN RFC NUMBER. AND THERE ARE FOUR CATEGORIES OF RFCS, SO NOT ALL RFC ARE STANDARD DOCUMENTS. THEY ARE -- THE CATEGORIES ARE: INFORMATIONAL, EXPERIMENTAL, STANDARDS TRACK, AND HISTORIC. IF YOU GET AN RFC ON EACH -- ON THE FIRST PAGE OF EACH DOCUMENT THE CATEGORY OF THE DOCUMENT IS CLEARLY MARKED. IT WILL SAY INFORMATIONAL, EXPERIMENTAL, STANDARDS TRACK, OR HISTORIC. AND ONLY STANDARD TRACK DOCUMENTS ARE CONSIDERED IETF STANDARDS.

AND THERE ARE THREE LEVELS OF STANDARDS. THESE ARE NOT ON THE FACE OF THE RFC BUT THERE IS A WEBSITE WHERE YOU CAN GO TO LOOK UP THE STATUS. THEY ARE: PROPOSED STANDARD, DRAFT STANDARD, AND INTERNET STANDARD. SO HOW DO YOU ELEVATE STANDARDS FROM PROPOSED STANDARD, TO THE NEXT LEVEL, TO FINALLY THE INTERNET STANDARD? WELL, THE PROCESS IS PRETTY COMPLICATED AND NOT OBVIOUS A LOT OF TIMES. WHAT WE KNOW IS THAT A MAJORITY OF THE COMMONLY USED INTERNET STANDARDS, ABOUT 90 PERCENT OF THE STANDARDS AVAILABLE TODAY ARE A PROPOSED STANDARD. VARIOUS REASONS WHY THEY DON'T MOVE TO THE NEXT LEVEL.

SIP, WHICH IS THE PROTOCOL WE ARE TALKING ABOUT, HAS BEEN BROADLY DEPLOYED IN THE LAST TEN YEARS AND THAT IS STILL A PROPOSED STANDARD. SOMETHING THAT WE PROBABLY USE EVERY DAY

WHICH IS HTTP, THIS IS THE PROTOCOL THAT IS USED BETWEEN WEB BROWSERS AND WEBSITE. WHEN YOU ACCESS A WEBSITE, THAT IS A DRAFT STANDARD.

EVEN PROTOCOL THAT ARE INTERNET STANDARDS CAN BE MODIFIED. SO, FOR EXAMPLE, THIS IS ANOTHER PROTOCOL THAT WE PROBABLY USE ALL THE TIME WHICH IS TO DO EMAIL. THE TWO STANDARDS REQUIRED FOR THAT IS MAIL AND SMTP. THEY WERE ORIGINALLY RFC 821 AND 822. AND THESE DOCUMENTS REACHED INTERNET STANDARD STATUS. BUT NETWORK CHANGE AND THINGS EVOLVE SO THEY BECAME OBSOLETED BY RFC 5321 AND 5322, WHICH WHEN IT WAS PUBLISHED ORIGINALLY WAS PUBLISHED AS PROPOSED STANDARD AND CURRENTLY ARE DRAFT STANDARD.

SO, DOES IT MEAN THAT PROPOSED STANDARDS ARE NOT STANDARDS? NO. IN FACT, IF YOU PICK UP AN RFC DOCUMENT AS A CATEGORY. THE ONLY THING YOU SEE IS THAT IT'S A STANDARDS TRACK DOCUMENT. IT DOESN'T MEAN THAT THEY -- THE STANDARD DOCUMENT IS NOT A SOLID JUST BECAUSE THEY HAVE DIFFERENT LEVELS OF MATURITY.

GOING BACK TO THESE TWO ORGANIZATIONS AS WE TALKED ABOUT, ITU-T IS FOCUSED IN THE TELECOMMUNICATIONS INDUSTRY AND THE ARCHITECTURE HERE IS MUCH MORE TELECOM-CENTRIC. HERE WE HAVE THE MEDIA GATEWAY CONTROLLER, THUS THE BLUE BOX ON THE TOP. IT'S ALSO COMMONLY CALLED IN THE INDUSTRY AS A SOFTSWITCH. WHAT YOU SEE IN THIS ARCHITECTURE IS THAT THIS BOX PLAYS THE SAME ROLE AS A CLASSIFIED SWITCH.

THE COURT: AS A WHAT?

MS. SHARPER: CLASSIFIED SWITCH. THAT'S THE SWITCH IN THE CENTRAL OFFICE IN THE TRADITIONAL PSTN NETWORK. SO WHAT THAT MEANS IS THAT YOU CAN HAVE THE TELEPHONES CONNECT TO EACH OTHER WITHOUT THE MEDIA GATEWAY CONTROLLER. SO IT ACCOMPLISHED THAT BY USING THIS MEGACO PROTOCOL, WHICH IS KNOWN AS A MASTER-SLAVE PROTOCOL. WHAT DOES THAT MEAN? THAT MEANS THAT THE RESIDENTIAL GATEWAY WHICH ACTUALLY HAS A TELEPHONE ATTACHED TO IT DOESN'T DO ANYTHING UNLESS IT'S TOLD TO DO SO BY THE SOFTSWITCH USING THE MEGACO. SO IT GETS A COMMAND AND IT SENDS A RESPONSE. SO THAT IS THE TYPE OF ARCHITECTURE THAT IS PROMOTED BY MEGACO USING THE ITU-T STANDARD.

HERE WHAT YOU SEE EVEN THOUGH IT'S USING IP FOR TRANSPORT, THE NETWORK ARCHITECTURE IS PRETTY MUCH THE SAME AS PSTN NETWORK, WHERE THE SMARTS ARE IN THE NETWORK, THE INTELLIGENCE AND CONTROL, ARE ALL IN THE GATEWAY CONTROLLER WITHIN THE NETWORK ITSELF. AND THE DEVICES ARE PRETTY DUMB. THEY JUST DO WHAT THEY ARE TOLD TO DO.

THIS IS VERY DIFFERENT FROM INTERNET-CENTRIC ARCHITECTURE THAT WAS BEING PROMOTED BY SIP. HERE YOU SEE TWO CIRCLES REPRESENTING CUSTOMER PREMISES, OR IN THIS CASE ON THE RIGHT IS A SMALL OFFICE. AND WHAT YOU SEE IS THAT THESE NETWORKS CAN TALK TO EACH OTHER USING SIP. YOU DON'T SEE SOMETHING THAT IS AN EQUIVALENT OF A CLASSIFIED SWITCH IN THE NETWORK.

THIS IS POSSIBLE BECAUSE THE DEVICES TALK TO EACH OTHER USING SIP WHICH IS A PEER-TO-PEER PROTOCOL AS OPPOSED TO A MASTER-SLAVE PROTOCOL.

SO, YOU DON'T NEED A CENTRALIZED SERVER ANYWHERE WITHIN THE NETWORK FOR THESE DEVICES TO TALK TO EACH OTHER. WHAT YOU SEE HERE IF YOU CAN THINK OF IT'S MORE LIKE HOW EMAIL WORKS. SO THE ARCHITECTURE HERE IS INTERNET-CENTRIC VERSUS HOW A TELEPHONE SYSTEM WILL WORK. ALTHOUGH THERE COULD BE OPTIONAL NETWORK-BASED SERVICES THAT CAN BE MADE AVAILABLE TO THE SIP DEVICES, THEY ARE NOT NECESSARY. SO WHAT PEOPLE TALK ABOUT IN THE INTERNET ENVIRONMENT IS THAT THE SMARTS ARE IN THE END-USER DEVICES AND THE NETWORK JUST SERVE AS A TRANSPORT. IT'S JUST A WIRE, EXCEPT A LITTLE BIT MORE COMPLICATED THAN JUST A POINT-TO-POINT WIRE THAT CONNECTS THE DEVICES TO EACH OTHER.

HERE IS A TIMELINE FOR SIP DEVELOPMENT. I WON'T GO THROUGH A LOT OF DETAILS THAT IS IN THIS SLIDE. SUFFICE IT TO SAY THAT THE ORIGINAL SPECIFICATION THAT WERE INTERNET DRAFTS WERE PROPOSED IN FEBRUARY OF 1996. AND OFFICIAL INTERNET DRAFTS WAS SUBMITTED TO IETF LATER THAT YEAR, AND IT WENT THROUGH OVER TWO YEARS OF DEVELOPMENT WITHIN THE WORKING GROUP BEFORE EVERYBODY WAS SATISFIED WITH THE PROTOCOL, AND IT WAS FINALLY SUBMITTED TO THE ENGINEERING STEERING GROUP FOR APPROVAL. AND THE OFFICIAL APPROVAL FOR THE PROTOCOL WAS RECEIVED IN 1999.

AFTER THAT APPROVAL, THE INDUSTRY WORKED HARD AT

DEVELOPING PRODUCTS THAT WERE INTEROPERABLE. THERE WERE MULTIPLE INTEROPERABILITY EVENTS. AND BY 2001, THESE EVENTS WERE ATTENDED BY MORE THAN 50 COMPANIES TRYING TO MAKE THEIR EQUIPMENT WORK TOGETHER. AND IN 2001, THE FIRST VOICE OVER IP SERVICE USING SIP WAS LAUNCHED BY VONAGE. SO BY THEN THE SIP-BASED NETWORK ARCHITECTURE WAS IN PLACE.

THESE ARE SOME OF THE COMMENTS THAT PEOPLE MADE ABOUT SIP IN THAT TIME FRAME. HENRY SINNREICH, WHO WAS ASSOCIATED WITH WORLDCOM, MADE THE COMMENT THAT SIP IS THE PURE INTERNET PLAY AS OPPOSED TO THE TRADITIONAL TELECOM NETWORK. THIS IS WHAT AN INTERNET APPLICATION LOOKS LIKE. MATT JOHNSON FROM LEVEL 3 SAID: BY PUSHING INTELLIGENCE OUT TO THE NETWORK ENDPOINTS, SIP CAN SIMPLIFY THE SYSTEMS REQUIRED IN THE NETWORK, AGAIN ACKNOWLEDGING THE DIFFERENCE OF SIP FROM OTHER PROPOSED PROTOCOLS. AND VINT CERF, WHO IS OFTEN CALLED THE FATHER OF THE INTERNET, SAID: SIP IS PROBABLY THE THIRD GREAT PROTOCOL OF THE INTERNET, AFTER TCP/IP AND HTTP. AND THIS WAS IN 2001.

SO, OKAY, WHAT IS SIP? AS SPECIFIED BY RFC 2543, IT'S A SIGNALING PROTOCOL FOR CREATING, MODIFYING, AND TERMINATING VOICE OVER IP CALLS. SO HOW DO YOU USE SIP TO BUILD A VOICE OVER IP NETWORK? IN ORDER TO EXPLAIN HOW YOU USE SIP TO BUILD A VOICE OVER IP NETWORK, RFC 2543 DEFINES A SET OF BUILDING BLOCKS AND THE BEHAVIOR AND THE PROTOCOL OF THESE BUILDING BLOCKS IN A VOICE OVER IP NETWORK. THERE ARE FIVE DIFFERENT TYPES OF BUILDING BLOCKS THAT ARE SPECIFIED IN RFC 2543. THEY

ARE: USER AGENT, PROXY SERVER, REDIRECT SERVER, LOCATION SERVER, AND REGISTRAR.

THE `519 PATENT ONLY TALKS ABOUT USER AGENT AND PROXY SERVER SO THAT'S WHAT WE WILL FOCUS IN THE TUTORIAL TODAY. EACH OF THESE BUILDING BLOCKS COMMUNICATE WITH EACH OTHER USING SIP MESSAGES. THERE ARE TWO TYPES OF MESSAGES: REQUEST MESSAGES AND RESPONSE MESSAGES. AND WE WILL TAKE A LOOK AT THIS A LITTLE BIT LATER IN THE TUTORIAL AS WELL. RFC 2543 IS CONCERNED WITH INTEROPERABILITY. IT'S NOT CONCERNED ABOUT HOW YOU USE THESE BUILDING BLOCKS. AT THE END OF THE DAY, THESE BUILDING BLOCKS ARE JUST SOFTWARE MODULES. SO HOW YOU USE THEM TO PUT A PRODUCT TOGETHER IS NOT OF CONCERN IN THE SPECIFICATION. IT'S MORE THE PROTOCOLS AND HOW THESE BUILDING BLOCKS INTERACT WITH EACH OTHER. THAT'S WHAT IS SPECIFIED IN THE RFC ITSELF.

OKAY, HERE IS A PICTORIAL REPRESENTATION OF A SIP USER AGENT. THE DEFINITION FOR USER AGENT SAYS: IT'S A PROGRAM THAT CONTAINS BOTH A USER AGENT CLIENT AND A USER AGENT SERVER. OKAY, SO WHAT DOES THAT MEAN? IT THEN DEFINES USER AGENT CLIENT AS A PROGRAM THAT INITIATES THE SIP REQUEST, AND A USER AGENT SERVER AS A PROGRAM THAT CONTACTS A USER WHEN THE SIP REQUEST IS RECEIVED AND RETURNS THE RESPONSE ON BEHALF OF THE USER. THE RESPONSE ACCEPTS, REJECTS, OR REDIRECTS THE REQUEST.

THE DEFINITION ITSELF IS REALLY NOT SUFFICIENT FOR ONE OF

SKILL IN THE ART TO FIGURE OUT HOW YOU BUILD A SIP USER AGENT. IN ORDER FOR YOU TO KNOW HOW TO DO THAT, YOU'D NEED TO KNOW WHAT THESE SIP REQUEST AND RESPONSE MESSAGES ARE AND HOW THE SIP USER AGENT ACTUALLY USES THEM. SO THESE ARE COVERED IN DIFFERENT SECTIONS OF THE RFC. THE PROTOCOLS, THE MESSAGE FORMAT ARE SPECIFIED IN SECTIONS 2 THROUGH 7 OF THE DOCUMENT. AND THE RULES ASSOCIATED WITH HOW A SIP USER AGENT SENDS AND RECEIVES THESE MESSAGES ARE COVERED IN SECTION 11. WE WILL TAKE A CLOSER LOOK AT HOW THIS IS DONE IN A MINUTE.

HERE IS A PICTORIAL REPRESENTATION OF A SIP PROXY SERVER. ACCORDING TO THE DEFINITION, IT'S AN INTERMEDIARY PROGRAM THAT ACTS AS BOTH A SERVER AND A CLIENT FOR THE PURPOSE OF MAKING REQUESTS ON BEHALF OF OTHER CLIENTS. SO WHAT DOES THIS MEAN? IN THE TERMINOLOGY OF RFC 2543, A SERVER IS NOT A COMPUTER WHICH IS WHAT WE WOULD NORMALLY THINK OF. A SERVER IS A SOFTWARE PROGRAM THAT ACCEPTS SIP REQUESTS AND SENDS A RESPONSE TO IT. SO THAT'S A SERVER. AND THAT'S SHOWN IN THE PICTURE ON THE LEFT. SO, SIP PROXY SERVER ACCEPTS A RESPONSE AND SENDS A REQUEST BACK.

THE OTHER HALF OF A SIP PROXY SERVER IS A CLIENT. AND THE DEFINITION OF A CLIENT WITHIN THE SPEC IS IT'S ANYTHING THAT SENDS SIP REQUESTS. THAT'S A CLIENT. SO WE HAVE TO READ A LITTLE BIT MORE TO SEE WHAT THIS THING DOES. REQUESTS ARE SERVICED INTERNALLY OR BY PASSING THEM ON, POSSIBLY AFTER TRANSLATION, TO OTHER SERVERS. SO A SIP PROXY SERVER NORMALLY

IS ASKED TO FIND A LOCATION OF A SIP USER AGENT. IN ORDER TO DO THAT, A SIP PROXY SERVER MAY CONSULT A LOCATION SERVER WHICH HAS THE CALLED PARTY'S LOCATION INFORMATION. IN THAT CASE, IT CAN RESOLVE THE ADDRESS AND SENDS A SIP REQUEST ONTO THIS USER AGENT. SOMETIMES IT CONSULTS ITS LOCAL LOCATION SERVER AND IT DOESN'T HAVE THE ADDRESS. IN THAT CASE, THE REQUEST IS FORWARDED TO ANOTHER SERVER.

AGAIN, HOW THIS IS DONE IS NOT IN THE DEFINITION. THE RULES ASSOCIATED WITH HOW YOU IMPLEMENT THE PROXY SERVER IS FOUND IN SECTION 12.3 OF THE DOCUMENT. BUILDING A SIP USER AGENT AND SIP PROXY SERVER REQUIRES YOU TO USE SIP REQUEST MESSAGES. AND THIS IS SOMETHING THAT IS IN SECTION 4 OF THE SPECIFICATION. LET'S REVIEW WHAT ARE THE SIP REQUEST MESSAGES THAT ARE AVAILABLE. THEY ARE: INVITE, WHICH IS USED TO INVITE A USER TO A COMMUNICATION SESSION; ACK, WHICH ACKNOWLEDGES THE FINAL RESPONSE TO AN INVITE REQUEST; OPTIONS, WHICH IS USED TO QUERY A SERVER ABOUT ITS ABILITIES; BYE, TO TERMINATE A CALL SESSION; CANCEL, TO CANCEL A PENDING REQUEST; AND REGISTER, TO REGISTER AN ADDRESS WITH A SIP REGISTRAR.

THE DETAIL OF THE SYNTAX OF THESE MESSAGES ARE IN SECTION 4 TO 6, BUT LET'S AT LEAST TAKE A LOOK AT ONE EXAMPLE OF AN INVITE REQUEST MESSAGE. SO WHAT YOU SEE IN THE FIRST LINE IS THAT THE MESSAGE STARTS WITH A KEY WORD, IN THIS CASE THE WORD INVITE. THE SECOND FIELD IN THIS FIRST LINE IS THE ADDRESS OF THE CALLED PARTY. THIS LOOKS KIND OF SIMILAR TO AN EMAIL

ADDRESS BECAUSE, AS I SAY, A SIP IS PRETTY CLOSE TO HOW EMAIL WORKS. SO HERE YOU HAVE THE ADDRESS AS SIP:USERB@THERE.COM. SO THAT WOULD BE THE FIELD THAT TELLS A SERVER WHO TO INVITE.

AND THAT'S FOLLOWED BY THE VERSION NUMBER OF SIP. HERE IT SAYS IT'S VERSION 2.0. I WON'T GO INTO EVERY LINE OF THIS REQUEST MESSAGE, BUT I WOULD LIKE TO NOTE THAT EVERY REQUEST MESSAGE HAS TO HAVE THE FROM, TO, AND CALL-ID HEADERS. THESE ARE IMPORTANT BECAUSE THEY ARE NECESSARY TO IDENTIFY A SPECIFIC CALL SESSION. SO THIS IS A PACKET NETWORK AND YOU NEED TO KNOW WHICH MESSAGE IS ASSOCIATED WITH WHICH CALL, SO THESE FIELDS ARE USED FOR THAT PURPOSE.

THERE ARE ALSO SIX DIFFERENT TYPES OF SIP RESPONSE MESSAGES AND THAT'S SHOWN HERE. SIP RESPONSE MESSAGES HAS A DIFFERENT FORMAT THAN WHAT WE PREVIOUSLY SAW. RATHER THAN ONE KEY WORD, IT ACTUALLY STARTS WITH A THREE-DIGIT STATUS CODE. AND THAT STATUS CODE IS THEN FOLLOWED BY A REASON PHRASE, AN ENGLISH EXPLANATION OF WHAT THE MESSAGE MEANS. SO THE FIRST TYPE OF STATUS CODE STARTS WITH THE DIGIT 1 THAT'S SHOWN ON THE TABLE IN THE RIGHT. SO IT'S, FOR EXAMPLE, 180 WHICH MEANS RINGING. THIS TYPE OF CODE IS USED TO PROVIDE INFORMATION. SO THE REQUEST HAS BEEN RECEIVED AND IT'S PROCESSING THE REQUEST, AND IN THE MEANTIME, IT TELLS THE OTHER PARTY SOME INFORMATION ABOUT THE CALL.

THE SECOND TYPE ARE KNOWN AS SUCCESS-TYPE MESSAGES WHICH MEANS THAT THE ACTION WAS SUCCESSFULLY RECEIVED AND ACCEPTED.

THE THIRD TYPE OF MESSAGES ARE REDIRECTION MESSAGES WHICH SAYS THAT A FURTHER ACTION NEEDS TO BE TAKEN IN ORDER TO COMPLETE THE REQUEST. THEN THERE ARE DIFFERENT TYPES OF ERROR MESSAGES. IF THE ERROR WAS CAUSED BY A CLIENT, IT STARTS WITH STATUS CODE OF 4; SERVER ERROR 5; GLOBAL ERROR 6.

HERE IS WHAT A RESPONSE MESSAGE LOOKS LIKE. SO THE FIRST LINE STARTS WITH A SIP PROTOCOL VERSION RATHER THAN INVITE OR KEY WORD, FOLLOWED BY THE STATUS CODE AND THE REASON PHRASE. IN THIS CASE IT SAYS: 180 RINGING. WHAT YOU SEE IS THAT THE FORMAT FOLLOWING THE FIRST LINE IS VERY SIMILAR TO REQUEST MESSAGES. AND THE IMPORTANT THING TO NOTE HERE IS THAT AGAIN THE FIELD FROM, TO, CALL-ID ARE MANDATORY BECAUSE THEY ARE NECESSARY TO IDENTIFY A SPECIFIC CALL SESSION.

SO HOW ARE THESE MESSAGES USED BY THESE BUILDING BLOCKS? WE SAW THIS DIAGRAM OR SOMETHING SIMILAR A LITTLE WHILE AGO. BUT HERE WHAT WE ARE SHOWING IS COMMUNICATION HAPPENING DIRECTLY BETWEEN ONE USER AGENT TO ANOTHER USER AGENT. USER AGENT ON THE LEFT IS INITIATING THE CALL SESSION BY SENDING AN INVITE. THE USER AGENT ON THE RIGHT SENDS AN INFORMATIONAL RESPONSE SAYING 180 RINGING. WHEN THE USER ACTUALLY ANSWERS THE PHONE, IT SENDS 200 OK SAYING THAT NOW WE CAN START THE COMMUNICATION. THE USER AGENT RESPONDS WITH AN ACK. THAT MEANS THAT IT RECEIVED A RESPONSE AND COMMUNICATION CAN HAPPEN BETWEEN THE TWO. AND WHEN SOMEBODY HANGS UP THE PHONE, THE USER AGENT SENDS THE BYE MESSAGE.

THIS IS WHAT HAPPENS WHEN THE ORIGINATING USER AGENT KNOWS THE LOCATION OF THE RECEIVING USER AGENT ON THE RIGHT. SOMETIMES, THOUGH, A USER AGENT MAY WANT TO CONTACT SOMEONE EVEN THOUGH IT DOESN'T KNOW THE LOCATION OF THAT PERSON. SO THAT'S THE CASE WHEN A PROXY SERVER GETS INVOLVED. HERE, THE SIP USER AGENT IS SENDING AN INVITE MESSAGE BUT DOESN'T REALLY KNOW WHERE THE USER AGENT THAT IT WANTS TO REACH IS LOCATED. SO RATHER THAN SENDING IT DIRECTLY, IT SENDS TO A SIP PROXY SERVER.

NORMALLY, A SIP USER AGENT GETS CONFIGURED WITH A DEFAULT PROXY SERVER SO IT WILL SEND THAT MESSAGE OVER THERE. THE SIP PROXY SERVER SENDS A MESSAGE BACK SAYING, OKAY, IT'S GOING TO TRY TO FIND THIS PERSON. AND IF IT FINDS IT, THEN IT FORWARDS THE REQUEST ON TO THE DESTINATION USER AGENT. AND FROM THERE ON, IT SERVES AS INTERMEDIARY OR A RELAY POINT BETWEEN THESE MESSAGES. SO THE MESSAGES GOES BACK AND FORTH THROUGH THIS PROXY SERVER. SO SOMETIMES THIS PROXY SERVER DOESN'T HAVE THE ADDRESS, SO WHAT DOES IT DO? IT CAN SEND IT TO ANOTHER PROXY SERVER. THIS IS AN EXAMPLE WHERE TWO PROXY SERVERS ARE INVOLVED IN THE PASS FROM SIP USER AGENT ON THE LEFT TO THE RIGHT.

SO LET'S TAKE A CLOSER LOOK AT WHAT'S ACTUALLY HAPPENING. THIS IS THE INVITE MESSAGE FROM USER AGENT THAT IS BEING SENT ORIGINALLY. THIS IS THE SAME MESSAGE THAT WE SAW BEFORE. IT'S TRYING TO REACH USERB@THERE.COM. SO THIS MESSAGE IS SENT

TO THE SIP PROXY SERVER. HERE IS THE INVITE MESSAGE FROM THE SIP PROXY SERVER GENERATED BY THE FIRST SIP PROXY SERVER. WHAT I'VE DONE IS HIGHLIGHTED IN RED THE TWO ADDITIONAL LINES THAT IS INSERTED. YOU SEE IT'S PRETTY MUCH A COPY OF WHAT IT RECEIVED, EXCEPT WHAT IT DOES IS THAT IT ADDS SOME ROUTING INFORMATION, WHICH IS THE ADDRESS INFORMATION OF THE PROXY SERVER ITSELF SO THE MESSAGE CAN FIND ITS WAY BACK TO IT.

THE CALL SESSION IDENTIFICATION INFORMATION IN THIS CASE REMAINS THE SAME BECAUSE IT'S THE SAME CALL. HERE IS WHAT THE SIP PROXY SERVER, THE SECOND ONE, WHAT THE INVITE MESSAGES LOOKS LIKE. THE TWO ADDITIONAL LINES THAT ARE INSERTED IS IN AQUA BLUE. WHAT YOU SEE AGAIN IS THAT IT'S REALLY JUST ADDING THE ROUTING INFORMATION. AND THE CALL SESSION IDENTIFICATION REMAINS THE SAME. AGAIN, THIS ROUTING INFORMATION IS NECESSARY BECAUSE THE MESSAGE AS IT FINDS ITS WAY BACK HAS TO GO THROUGH BOTH OF THESE PROXY SERVERS TO REACH THE ORIGINATING SIP USER AGENT.

OKAY, SO THAT'S KIND OF A TUTORIAL ON SIP, AND LET'S TAKE A LOOK AT THE '519 PATENT. THIS IS JUST A NOTE THAT THESE PATENTS WERE FILED AFTER 2000 WHEN ALL THE STANDARDS WERE ALREADY IN PLACE.

HERE IS THE FIRST FIGURE IN THE PATENT LABELED PRIOR ART, AND THIS IS DEPICTING THE TRADITIONAL PSTN NETWORK. AND MR. MCANDREWS COVERED THIS PRETTY WELL, SO I WILL SKIP THIS.

FIGURE 2 IS WHAT IS CALLED WITHIN THE PATENT THE NEXT

GENERATION NETWORK. IT'S BASED ON THE MEDIA GATEWAY CONTROLLER USING MEGACO PROTOCOL TO CONTROL THE RESIDENTIAL GATEWAYS TO SET UP CALL CONNECTION. THIS, HOPEFULLY, LOOKS FAMILIAR TO YOU BECAUSE IT'S REALLY THE SAME NETWORK ARCHITECTURE AS WHAT WE REVIEWED WHEN WE TALKED ABOUT THE ITU-T PROTOCOL-BASED ARCHITECTURE.

THAT PICTURE IS SHOWN AGAIN HERE, AND WHAT YOU SEE IS THAT THEY BOTH HAVE THE MEDIA GATEWAY CONTROLLER, THE RESIDENTIAL GATEWAYS, THE PSTN GATEWAY, AND THEY USE THIS MASTER-SLAVE MEGACO PROTOCOL TO COMMUNICATE BETWEEN THEM.

HERE IS FIGURE 3 OF THE PATENT AND IT'S CALLED THE EDGE SWITCH NETWORK ARCHITECTURE. WHAT THE PATENT SAYS IS THAT THERE ARE THREE CONNECTIVITY ELEMENTS IN THIS ARCHITECTURE: THE EDGE SWITCHES, WHICH ARE THE SWITCHES ON THE CUSTOMER PREMISES, THE APPLICATION SERVER, AND THE PSTN GATEWAY. APPLICATION SERVER ON TOP LEFT AND PSTN GATEWAY KIND OF IN THE MIDDLE OF THE DIAGRAM. AND ALL THESE CONNECTIVITY ELEMENTS USE SIP FOR COMMUNICATION AND THEY COULD OPTIONALLY CONNECT TO NETWORK-BASED SIP PROXY SERVERS. HERE YOU SEE THE SMARTS, AND THE NETWORK IS AT THE EDGE AND THE NETWORK ITSELF IS JUST A DUMB TRANSPORT NETWORK.

AGAIN, THIS IS PRETTY SIMILAR TO WHAT WE'VE SEEN BEFORE IN TERMS OF THE ARCHITECTURE USING SIP. HERE ARE THE SIP EDGE DEVICES WHICH ARE INTELLIGENT DEVICES THAT CAN USE SIP TO COMMUNICATE. HERE IS THE PSTN GATEWAY AND THE APPLICATION

SERVER, AND SIP USED BETWEEN THESE.

LET'S TAKE A CLOSER LOOK AT HOW THESE BUILDING BLOCK ELEMENTS ARE IMPLEMENTED WITHIN THE '519 PATENT. HERE IS FIGURE 7 OF THE PATENT WHICH DEPICTS THE SOFTWARE ARCHITECTURE OF THIS DEVICE. WHAT THE EDGE SWITCH DOES IS IT CONNECTS TWO TYPES OF TELEPHONES. ONE IS KNOWN AS THE SIP PHONE. THESE ARE PHONES A AND C, AND THE OTHER ONE IS A NON-SIP PHONE, PHONE B. THERE ARE SIP USER AGENTS OUTSIDE OF THIS BOX THAT'S NOT IN THE DIAGRAM, SO I'VE JUST PUT THAT IN THIS PICTURE TO ILLUSTRATE THAT THEY ARE THERE SO THAT THEY KNOW HOW TO SEND AND RECEIVE SIP MESSAGES.

SO LET'S TAKE A LOOK AT WHAT THE EDGE SWITCH DOES FOR A SIP-BASED PHONE. HERE WHAT YOU SEE IS THAT IT REALLY JUST SERVES AS A SIP PROXY SERVER. THE USER AGENT IN PHONE A SENDS A SIP REQUEST TO THE SIP PROXY SERVER WHICH IS INSIDE THE EDGE SWITCH. THE SIP PROXY SERVER THEN FORWARDS IT ON TO A SIP USER AGENT THAT REPRESENTS PHONE C. SO THAT'S THE FUNCTIONALITY THAT IS PROVIDED BY THE EDGE SWITCH.

FOR NON-SIP PHONES, THE EDGE SWITCH PROVIDES ONE ADDITIONAL FUNCTION WHICH IS A TELEPHONE GATEWAY FUNCTIONALITY. THIS IS SHOWN IN THE BLUE BOX. SO WHAT IT DOES IS THAT WHEN YOU PICK UP A TELEPHONE, THE SOFTWARE WITHIN THE BOX DETECTS THAT THERE WAS AN OFF-HOOK SIGNAL. AND IT SENDS A DIAL-TONE DOWN SO THE PHONE -- YOU WILL KNOW THAT YOU ARE CONNECTED. IT THEN USES THE SIP USER AGENT SOFTWARE THERE

TO CONVERT THESE EVENTS TO SIP MESSAGES. SO THE SIP USER AGENT WHEN THERE IS AN OFF-HOOK WILL START AN INVITE MESSAGE.

THAT INVITE MESSAGE THEN IS SENT TO A SIP PROXY SERVER WHICH HAPPENS TO BE IN THE SAME BOX. THESE ARE SOFTWARE BUILDING BLOCKS, SO YOU JUST LOAD TWO SOFTWARE MODULE INTO THIS SAME MACHINE. AND SIP PROXY SERVER TAKES A LOOK AT THIS AND FORWARDS IT TO THE SIP USER AGENT, AS WE TALKED ABOUT BEFORE. SO THIS IS HOW THE '519 EDGE SWITCH USES SIP BUILDING BLOCKS IN ITS PRODUCT. SO THAT IS THE END OF MY TUTORIAL.

THE COURT: THANK YOU.

MS. SHARPER: THANK YOU.

THE COURT: YOUR COMMENTS HAVE BEEN MOST HELPFUL. WE WILL START CLAIM CONSTRUCTION, TRY TO, PROMPTLY AT 9. DO YOU HAVE ANY COMMENTS?

MR. VERHOEVEN: YOUR HONOR, WE AGREED ON A MANNER OF PRESENTATION. I DON'T KNOW IF YOU WANT TO HEAR THE DETAILS OR NOT.

THE COURT: I WAS GOING TO TAKE UP IF I HAD ADDRESSED THAT IN MY SCHEDULING ORDER, WHICH I HAVE STARTED TRYING TO DO IN RECENT CASES. HOW DO THE PARTIES ANTICIPATE GOING FORWARD?

MR. VERHOEVEN: WELL, WE THINK THAT IF WE GO TERM BY TERM FOR EVERY TERM THAT WE WILL HAVE TROUBLE BECAUSE THERE'S SO MANY TERMS, AND IT MAY BE MORE EFFICIENT TO GROUP THE TERMS INTO THREE TRANCHES. AND THE PLAINTIFF WILL GO AND COVER THE FIRST TRANCHE, AND THEN I'LL GO AND THEN --

THE COURT: REPLY.

MR. VERHOEVEN: EXACTLY. AND WE'LL DO THAT, SO WE'LL BE UP IN FRONT OF YOU THREE TIMES EACH.

THE COURT: THAT'S AGREEABLE. I LIKE EITHER TERM OR GROUPS OF TERMS RATHER THAN ONE PARTY MAKING ALL THEIR PRESENTATION AND THEN A RESPONSE AND THEN A REPLY. SO MUCH EASIER TO FOLLOW, PARTICULARLY FROM THE WRITTEN TRANSCRIPT STANDPOINT, IF YOU GO FORWARD IN THE FASHION YOU HAVE SUGGESTED.

MR. VERHOEVEN: THANK YOU, YOUR HONOR.

THE COURT: PLEASE TRY NOT TO SPEND A LOT OF TIME ON THE GENERAL PRINCIPLES OF CLAIM CONSTRUCTION. I'VE HEARD THAT A FEW TIMES, AND IF I HAVEN'T LEARNED IT TO DATE, I'M PROBABLY NOT GOING TO. NOW, NEEDLESS TO SAY, IF YOU HAVE SOME CASE LAW THAT'S UNIQUE TO SOME OF THE ISSUES IN THIS CASE, DON'T HESITATE TO BRING IT TO MY ATTENTION, AND I'M SURE YOU HAVE IN THE BRIEFING. BUT AS FAR AS USING A LOT OF YOUR TWO HOURS ON GENERAL PRINCIPLES OF CLAIM CONSTRUCTION, TRY NOT TO DO THAT.

I NOTICE I GAVE EACH SIDE TWO HOURS RATHER THAN THE CUSTOMARY NINETY MINUTES. I AM ASSUMING AT THE SCHEDULING CONFERENCE THE PARTIES INSISTED THEY NEEDED MORE TIME. IS THAT CORRECT? I HAVE YOU DOWN TWO HOURS A SIDE. I HOPE I HAVEN'T TOLD YOU SOMETHING THAT YOU WERE NOT AWARE OF.

MR. MCANDREWS: NO, THAT'S TRUE, YOUR HONOR, BUT WE ARE HOPEFUL THAT NINETY MINUTES WILL BE SUFFICIENT.

THE COURT: NINETY. THEN MY PLANS WILL BE START PROMPTLY AT 9:00 AND WE'LL GO UNTIL WE FINISH RATHER THAN TAKING A LUNCH BREAK. IF YOU PLANNED ON USING THE ENTIRE FOUR HOURS, I'D PROBABLY GO TO ABOUT NOON AND TAKE A SHORT LUNCH BREAK AND THEN COME BACK FOR THE BALANCE. BUT IF BOTH SIDES ANTICIPATE SOMETHING LIKE NINETY MINUTES, THEN WE'LL PROBABLY JUST GO RIGHT THROUGH THE --

MR. VERHOEVEN: THAT'S FINE, YOUR HONOR.

THE COURT: -- NINETY -- THREE HOURS. ANYTHING ELSE?

MR. VERHOEVEN: NO, YOUR HONOR.

THE COURT: THEN WE'LL TRY TO START PROMPTLY AT 9:00. WE WILL SEE EVERYONE TOMORROW.

(ADJOURNED AT 3:23 P.M.)

REPORTER'S CERTIFICATION

I CERTIFY THAT THE FOREGOING IS A CORRECT TRANSCRIPT FROM THE RECORD OF PROCEEDINGS IN THE ABOVE-ENTITLED MATTER.

DATED THIS 18TH DAY OF JUNE, 2009.

/S/LIBBY CRAWFORD

LIBBY CRAWFORD, CSR

OFFICIAL COURT REPORTER