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Understanding Voice over IP Signaling Protocols in Cisco Telephony Implementations

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Introduction

Understanding the specific functions Voice over Internet Protocol (VoIP) signaling protocols perform plays a critical role in designing, building, maintaining, and troubleshooting a Cisco VoIP implementation. Each protocol also plays a direct role in implementing different features, services, technologies, as well as in product selection and integration. Specifically, Cisco has predominately implemented the following five different signaling protocols amongst their rapidly expanding telephony product line:

- H.323
- MGCP (Media Gateway Control Protocol)
- Skinny (SCCP, Skinny Client Control Protocol)
- SIP (Session Initiation Protocol)
- CTI/JTAPI (Computer Telephony Integration/Java Telephony Application Programming Interface)

What is a signaling protocol?

In a very simple sense, all telephones perform two basic functions; signaling and audio. When a user picks up a phone and hears the dial tone, dials digits, another phone rings, put another on hold, create a conference call: this is signaling. When you talk on a phone to a person, in a conference call, or are leaving or hearing voice mail, even hearing Music on Hold (MOH): this is audio.

Traditional telephony, prior to the exponential growth of VoIP, performed signaling in either an analog or digital formats that still play an important role in voice gateways. While VoIP signaling protocols vary from vendor to vendor, some proprietary, others industry standard, audio is handled by one industry standard IP protocol called Real Time Protocol (RTP). Inside, an RTP frame contains the digitized conversion of human analog voice created from codecs (Coder-Decoder or Compress-Decompress). When bandwidth is abundantly available, as in a LAN environment, the Cisco IP phone default G.711 is used in RTP. When bandwidth is at a premium, as on a slow WAN link, G.729 is used. While other codecs are supported on some Cisco products, they are falling out of favor with designers and, thus, Cisco products as well.

In VoIP, IP phones have absolute reliance on VoIP signaling protocols. Unlike a traditional POTS (plain old telephony service) phone at home that still works, even if the home loses power, IP phones can only function with a VoIP signaling protocol, and the ability to call anywhere needed, further relies on VoIP signaling protocols. Another role of the signaling protocol is potentially to hand off the dialed number to a call agent like CallManager or CallManager Express, which then passes off the call signaling to another signaling protocol, ultimately to locate a phone to ring. With the emergency service 911, along with any other mission-critical or revenue-generating calls, any intelligent IP telephony design and implementation has a critical reliance on VoIP signaling protocols.

Why Voice over IP, and what is behind its incredible growth?

A fair question from any organization with a significant existing investment in traditional PBX technology is why to upgrade to a VoIP system, especially since the initial capital investment of VoIP can be sizeable. The best answer is a very large potential reduction in telephony operating costs in the form of reduced toll charges to send IP signaling and audio over existing IP WAN links. This can occur in two forms, Toll Bypass and Tail End Hop Off (TEHO). Suppose that a call went from Chicago to San Francisco where you have another office through a WAN link, but the destination was to another vendor in San Francisco. TEHO takes advantage of the WAN link, but then the call goes through a San Francisco gateway incurring only a local San Francisco toll charge. It is very common for organizations to see a full return on investment (ROI) from VoIP in a few years, or even months, depending on call patterns.

Many additional benefits of VoIP, although potentially less tangible, have to do with all the many integrations of VoIP possible with Internet technologies such as a concierge service in a hotel; database lookups for customers; and information brought to the IP phone, such as weather, stock quotes, airport delays; etc.

Cisco VoIP Products

Cisco CallManager (CCM)

In 1998, Cisco acquired Selsius, which brought both CallManager and the IP phones into the telephony product line. CallManager is usually the heart of a Cisco VoIP implementation, as it is the “brain” of the call control protocols to all IP telephony devices. CCM v4.x and earlier are installed on a Microsoft Windows 2000 server and, in CCM v5.0, is installed as an appliance on a Linux server. Clustering multiple servers is used for fault tolerance and load balancing, and can support 30,000 phones/ cluster. A future release of CCM v6.x will offer identical features on either a Microsoft or a Linux platform, whereas, currently, CCM v4.x and 5.x have differences in feature support.

Cisco CallManager Express/Cisco Unity Express (CME/CUE)

Years after the Selsius acquisition, Cisco created CME, which is a unique router internetworking operating system (IOS) feature sets ability to control IP phones with Skinny. CME is designed for small branch offices or retail stores to replace key systems or small PBXs with a current maximum of 240 IP phones supported in the LAN. CME can operate totally independently of CCM, or can co-exist with signaling protocols in an organization. CUE is separate hardware located physically in a router, which is a server running voice mail on a Linux platform.

Cisco Unity

Voice mail in a larger environment is handled on a Microsoft Windows 2000 or 2003 server running Unity. Unified messaging, which integrates voice mail, email, and faxes in a single messaging program like Outlook, is also available.

Cisco Unified Contact Center Express (UCCX)

Traditional telephony systems implement Automatic Call Distribution (ACD) and Interactive Voice Response (IVR) in call center products. CCM and CME offer only very limited ACD and IVR features, not suitable for a call center with many agents. UCCX supports both ACD and IVR with agent software and integrates directly in a CCM environment. Future versions may support integration directly with CME.

VoIP Signaling Protocols in a Cisco Implementation

The VoIP industry has developed many different signaling protocols, some proprietary, some governed by the Internet Engineering Task Force (IETF), and another from the International Telecommunication Union (ITU). The five below, not in any particular order, are the predominant focus in a Cisco VoIP installation. Also important to note is that all of these protocols are based on IP, thus, requiring an existing IP infrastructure to operate. Any error in IP or relating to IP, such as convergence issues with Spanning tree protocol or the routing protocol or flapping WAN links, will adversely affect IP telephony. One feature of IP signaling protocols is call survivability. When an existing VoIP call has an established RTP stream, and then any failure in IP connectivity occurs between either phones and their signaling device, such as CCM or CME, if the call stays up, there is call survivability.

H.323

H.323, like any other telephony or non-telephony protocol with the name format "letter.number," is governed by the ITU. In the early 1990s, the ITU changed their name from the Consultative Committee on International Telephony and Telegraphy (CCITT [the full correct name is in French]). The origin of the CCITT actually dates back to the original wired communication systems emanating from the telephony pioneers Samuel Morse of the Morse code and Alexander Graham Bell of the first telephone. H.323 is, therefore, considered quite mature. Engineers familiar with ISDN Q.931 will note many functional similarities to H.323, as they were both developed by the ITU.

Cisco implements H.323 in several formats. If an organization has two or more CallManager clusters, the logical IP connection between the clusters configured is called an Inter Cluster Trunk (ICT), which is based on H.323. H.323 also can be used as a signaling protocol between CCM and a voice gateway. If the voice gateway were not a Cisco product, H.323 would be the choice. One drawback of H.323 is that it does not support call survivability.

If two or more voice gateways or CallManager Express routers need to send signaling to each other over an IP network, the router configuration commands called dial peers are configured, also based on H.323. Many retail stores, pharmacies, doctor offices, etc. are beginning to install large numbers of CMEs in each of their locations. The number of configured dial peers needed for full connectivity requires a full mesh, from the formula:

$$\text{Number of Dial Peers required} = \frac{N(N-1)}{2}$$

Where N is the number of CMEs, voice gateways (or even entire CCM clusters with ICTs). In a large organization with hundreds or thousands of locations, administering and updating dial peers can be a sizeable task. The solution to simplify management and troubleshooting is another optional H.323 component called a gatekeeper. In essence, a gatekeeper can perform one or two functions; centralized call routing or Call Admission Control (CAC).

Centralized call routing allows for all of the Gateways, CMEs, or CCMs to be configured with a single dial peer or ICT pointer to the gatekeeper. The gatekeeper contains configurations of summarized ranges of phone numbers found in all of the branch locations that it uses to route phone calls.

CAC serves the purpose of "protecting voice from voice." A single G.729 call in an RTP stream takes approximately 26 kbps, while G.711 takes roughly 86 kbps, depending on the layer 2 headers. As an example, a slow WAN connection of 300 kbps, taking into account signaling, would allow resources for roughly 10 G.729 calls,

not to mention any other user data. What if 12 calls were submitted? How about 20? By default, CCM and CME assume infinite bandwidth everywhere and can send calls through a link that might not properly support them. The calls will go through, but at an unacceptable degradation in voice quality will occur in the form of clipping (pieces of voice missing) and a large audible delay. The solution is CAC, which is usually done by a gatekeeper, CCM, or both. CAC defines the maximum calls that can go through a link and, once all those calls are reached, additional calls are blocked or, ideally, routed through the public switched telephone network (PSTN).

Cisco implements an H.323 gatekeeper in a router via IOS commands and a gatekeeper does not require any unique router hardware modules, unlike a gateway. While a gatekeeper can also act as a gateway, this is rarely done in large production environments. Gatekeepers can also be configured in a fault tolerant group of gatekeepers called a gatekeeper cluster, different from a CCM cluster.

MGCP

Media Gateway Control Protocol (MGCP) is an Internet standard governed by the IETF, and is relatively much newer than H.323. MGCP is a client server protocol and the "client" can be a voice gateway or a user client, controlled by the server component referred to as a call agent. In a Cisco VoIP installation, MGCP plays one critical role: the logical control of voice gateways by the call agent CCM. MGCP implies more configuration steps in CCM with a minimal gateway configuration making MGCP a centralized signaling protocol. Since CallManager Express is built to operate independently, it is not controlled directly by CallManager; therefore, MGCP is not supported on CME, as it is not needed. CCM does not have the ability to use MGCP to non-Cisco voice gateways.

Skinny

Skinny is not an acronym, but it is often referred to as Skinny Client Control Protocol (SCCP). Skinny is entirely Cisco proprietary and came from the 1998 Selsius acquisition. The main function of Skinny is to control IP phones from either a CCM cluster or CME. If a voice gateway is configured as Survival Remote Site Telephony (SRST) to give remote phones fault tolerance to CCM in the event of a WAN failure, the remote phones will also talk Skinny to the SRST router. Skinny is also the control protocol between CCM and Unity. In an environment with a lot of analog phones or faxes, Cisco has a few voice gateway products with the "VG" designation, which are also controlled by Skinny from CCM or CME.

SIP

Session Initiation Protocol (SIP) is an IETF standard like MGCP. Just 3 years ago SIP played a relatively small role in Cisco telephony, but today it is gaining ground fast in new and future Cisco telephony products. Starting with CCM v4.0, a SIP trunk can be configured to implement voice signaling to another CCM cluster, a CME router, or any other non-Cisco IP telephony product that supports SIP. Voice gateways and CMEs can also do SIP signaling in a dial peer to any other SIP device. One mandatory use of SIP in Cisco is between CME and the internally installed voice mail component CUE. A big change in CCM 5.0 is the option to replace Skinny with SIP as the phone signaling protocol. CME v4.0 also supports SIP to the phones. One key benefit of running SIP to the phones is that Cisco now allows support of many of the fast growing third-party SIP phones. Cisco's new Presence server gives users a greater ability to locate another user in the communication format that they have available. The Presence server requires SIP with CCM v 5.0.

CTI/JTAPI

Computer Telephony Integration dates back to the 1960s with the idea of blending together telephones and computers. When you make a call to your bank or credit card company, you may have noticed that the agent often knows about your specific account before you ask. CTI takes the calling number and associates it with an account in a database that the agent sees on a computer screen. Telephony Application Programming interface (TAPI) was developed in 1993 by Microsoft and Intel to allow PCs to transparently communicate to different PBXs. Sun Microsystems variation of TAPI with Java is used in Cisco Telephony. CTI and JTAPI are used as a team in the call control of CCM to Unified Contact Center Express in a call center environment to perform ACD and IVR.

VoIP Protocol Integration

While all these 5 protocols are separate entities, they can all work together, even in a single user call. CCM has the inherent ability to translate from one signaling protocol to another as needed. CME can do some translations with some basic configurations, except the unsupported Media Gateway Control Protocol (MGCP).

As an example, a user calls in from the PST N to a call center. A voice gateway accepts the call, and MGCP controls the call, which is processed by CCM. CCM uses CTI to pass the call to UCCX, which accepts the call with JTAPI and puts the call in a queue. When an agent becomes available, UCCX talks with CCM, which uses Skinny to make the agent IP phone ring. Let's say the agent then needs to forward the call to a specialist located outside the CCM cluster. The CCM cluster has a gatekeeper controlled ICT to another cluster that uses H.323 to pass the call to the gatekeeper and then the next CCM cluster. Finally, the remote CCM v5.x cluster uses SIP to make the remote phone ring. As a reminder, any time in this sequence there is human voice, RTP is used, not a signaling protocol.

HTTP on Phones

While Hyper Text Transfer Protocol (HTTP) is not a VoIP signaling protocol, it can be used to add functionality and desirable features on an IP phone. When a user presses the Service button on the phone, Skinny talks to CCM, but the actual web content, such as a phone list, airport delays, weather report or a hotel menu, is delivered with HTTP to the phone screen directly.

DTMF Relay

Dual Tone Multiple Frequency (DTMF) was developed in the early 1960s as a replacement to the old pulse or rotary phones to create unique analog sounds used for voice signaling in analog phones. As an example of DTMF relay, let's say a user on one IP phone calls another IP phone, which is not present, and the call goes to voice mail (or an IVR system). As the user is hearing the voice mail prompt as well as leaving the audio voice mail, RTP is used directly between the user phone and the IP voice mail itself. A problem arises when the user enters a DTMF digit in the voice mail session to "deliver with normal priority" as codecs like G.729 are designed for human voice, not DTMF digits. The solution is to configure DTMF relay, which removes the DTMF digits from RTP and puts them in the signaling protocol where they belong. DTMF relay is not required to be configured if CCM is used with Unity as Skinny has contains the DTMF digits inherently. Other signaling protocols require DTMF relay to be manually configured for voice mail and IVR sessions.

QoS and Bandwidth Issues of VoIP signaling

In terms of the impact of bandwidth on an existing IP network, human audio voice contains the vast majority of the traffic. IP designers usually account for 5% of the RTP bandwidth for signaling. Still, both audio and signaling require a high prioritization in router and switch queues, especially on slow WAN links. Cisco

automatically marks the RTP frames as Expedited Forwarding (EF), which needs to be configured to use Low Latency Queuing (LLQ) on router interfaces. Signaling protocols have been marked to Assured Forwarding (AF31), but Cisco is migrating this default to Class Selector (CS3).

Summary

In summary, the quantity of new Cisco VoIP installations continues to grow at an ever increasing pace. The potential of reduced operating costs as well as a wide array of added features continue to attract more companies and government agencies world wide.

The essence of a well operating VoIP installation is a clear understanding of the protocols involved along with a proper configuration on all of the integrated products.

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About the author

Chris Olsen is a CCSI and has contracted with Global Knowledge as a Cisco Instructor since Global Knowledge's acquisition of Geotrain in 1999. After teaching Novell classes at the MCNI/MCNE level and as an MCT teaching Microsoft MCSE classes in the 1990s, he taught Cisco CCNA, the entire Cisco CCNP track, BGP, CiscoWorks, and security. In recent years, he has become a SME in Cisco voice technologies and teaches the entire CCVP track along with IPTX, IPTD, UCCX and Unity. He has been an independent consultant since 1996. He holds a Bachelors of Science in Mechanical Engineering and a Masters of Science in Mechanical Engineering from Bradley University in Peoria, IL. Chris can be reached at chrisolsen@earthlink.net.