Voice over IP (VoIP)/SIP Infrastructure Considerations for the Interaction Center Platform®

Whitepaper

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Abstract

Using the Interaction Center Platform event processing technology and Platform-based Interactive Intelligence products designed for Internet Protocol (IP) telephony and the Session Initiation Protocol (SIP), organizations can leverage their existing infrastructure to effectively deploy voice over IP (VoIP) solutions This paper outlines various infrastructure/network considerations for deploying VoIP solutions via the Interaction Center Platform.

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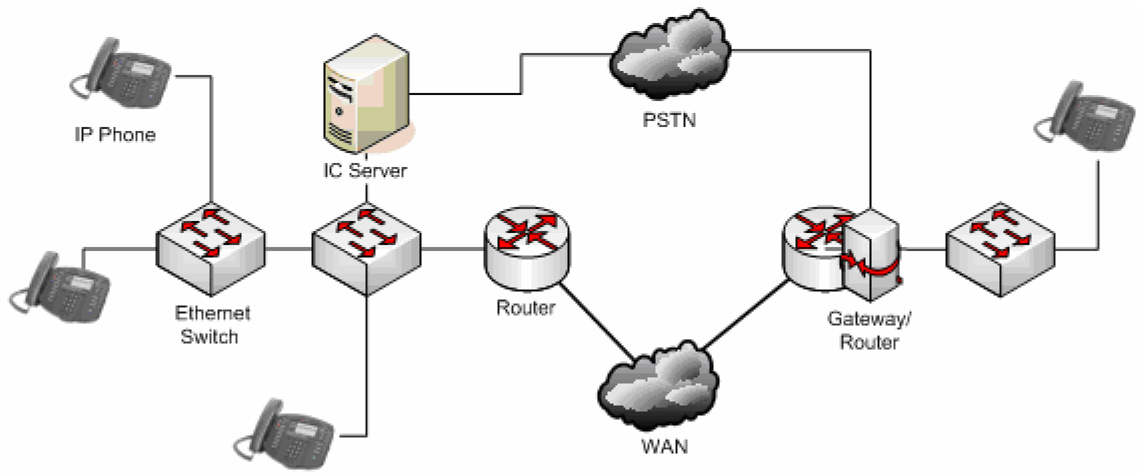
Interaction Center Platform Statement

The following Interactive Intelligence products are based on the Interaction Center Platform event processing technology and are designed for use in SIP-enabled voice over IP (VoIP) configurations.

- Customer Interaction Center® (CIC)
- Enterprise Interaction Center® (EIC)
- Communité®
- Vocalité™

The infrastructure considerations discussed in this white paper focus primarily on the *Customer Interaction Center* and *Enterprise Interaction Center* solutions.

Contents



Introduction

Since 2002 the Interaction Center Platform® event processing technology has enabled voice over IP (VoIP) communications utilizing the Session Initiation Protocol, or SIP. Interactive Intelligence introduced its VoIP/SIP capabilities in version 2.2 of the *Customer Interaction Center*® (CIC) software built on the Interaction Center Platform, and today offers the same IP capabilities in the Platform-driven *Enterprise Interaction Center*® (EIC) IP PBX software. (The *Communité*® (ka-mune-i-tay) unified communications software also comes SIP-enabled for VoIP.) Because CIC and EIC are both application-based solutions, deploying either of them in a VoIP configuration allows contact centers and enterprises to leverage an existing network infrastructure to converge communications and applications paths. "Converged" voice and data communications also reduce costs significantly over traditional PBX and multi-box hardware systems, especially for distributed multi-site organizations.

If your organization is considering the Interaction Center Platform technology for voice over IP and SIP, this paper addresses the many aspects of planning for VoIP and the factors that impact it, including these pre-deployment considerations:

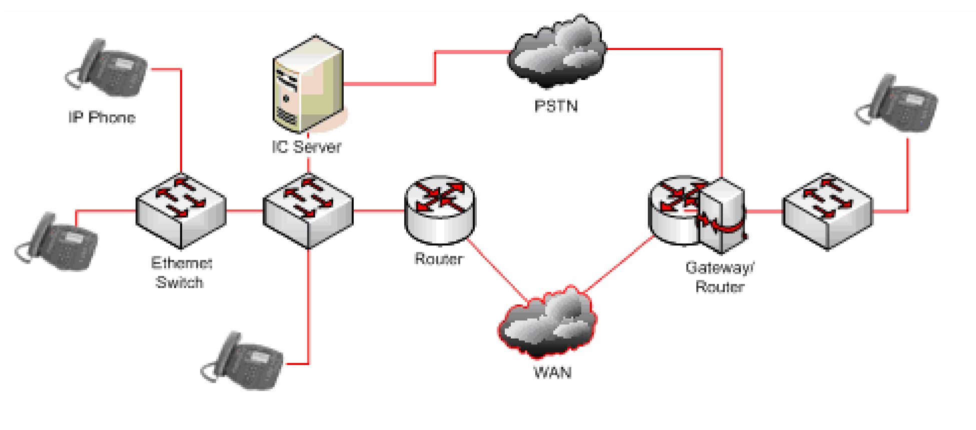
- **Network capacity (bandwidth):** Bandwidth utilization of voice devices.
- **Quality of Service (QoS):** Software for switches and routers to prioritize voice traffic.
- **Voice quality:** Factors that can affect voice quality.
- **Network devices:** Points to consider when examining your network devices.
- **Remote sites:** How remote sites might affect your network.
- **SIP-based Interaction Center Platform functional considerations:** How the Interaction Center utilizes SIP.
- **Network assessment:** Why a detailed network assessment is critical to the success of a voice network deployment.

Note that this paper is <u>not</u> intended as an analysis template to gauge network viability for a SIP-based VoIP deployment. The actual analysis of your organization's existing environment, and implementing a network topology into that environment, is a complex engineering process that should consider your network topology as well as the hardware vendor you choose, the call volumes you handle and the applications you implement. Interactive Intelligence (ININ) and or a certified ININ

Partner can assist with these engineering tasks in conducting a detailed network assessment for your organization.

Note as well that this paper is not intended to fully define SIP or the endpoints compatible with SIP deployments via the Interaction Center Platform technology and ININ's SIP-ready IP software solutions. Though Appendix A includes a brief overview of SIP, we urge you to visit the SipCenter at www.sipcenter.com to learn more about the Session Initiation Protocol, voice over IP, and the benefits these Internet-driven communications standards provide. Interactive Intelligence is a principal sponsor of SipCenter along with other SIP-based VoIP vendors.

VoIP Bandwidth Considerations



Considering how a SIP-based Interaction Center Platform solution operates — i.e., *Customer Interaction Center* (CIC), *Enterprise Interaction Center* (EIC) — it is important to factor in voice over IP bandwidth requirements. VoIP bandwidth is the combination of a variety of factors: the CODEC used (see ***Voice Compression*** next page), the underlying network path (Ethernet, frame relay, ATM), the number of calls being handled (call volume), and the position of the Interaction Center Server relative to end stations. Your organization must account for each of these factors when sizing a SIP-based Interaction Center VoIP network.

VoIP Call Volume

Call volume is determined the same way in packet- as well as circuit-switched voice networks. Many calculators are available to determine call volumes, such as Erlang.com (www.erlang.com). The biggest difference is, instead of acquiring a voice trunk, you will use the trunk amount as a multiple against which you apply your CODEC bandwidth.

Voice Path

Make sure to account for the actual audio path through every network aggregation point.

Voice Compression

Various network and phone vendors support a range of voice compression algorithms for assessing VoIP bandwidth. Among them is the Mean Opinion Score (MOS), a standard metric for quantifying the quality of voice compression techniques of different coder/decoders, or CODECs. Enacted by the International Telecommunications Union (ITU), MOS scores increase from 0 to 5 as the quality of voice communications improve. Table 1 shows how MOS increases as the required bandwidth for VoIP calls increases.

Compression Method	CODEC	Optimal Data Rate	MOS	Packetization Delay (msec)
PCM[1]	G.711	64 kbps	4.4	0.75
ADPCM	G.726	32 kbps	4.2	1
LD-CELP	G.728	16 kbps	4.2	3-5
CS-ACELP[1]	G.729	8 kbps	4.2	10
CS-ACELP*	G.729a	8 kbps	4.2	10
MPMLQ*	G.723.1	6.3 kbps	3.9	30
ACELP*	G.723.1	5.3 kbps	3.5	30

Table 1: Mean Opinion Scores (MOS)

Deployment Tip: Use the G.711 CODEC end-to-end, unless lack of capacity requires compression.

The G.711 CODEC offers the best voice quality, since it performs no compression, introduces the least delay, and is less sensitive than other CODECs to packet loss. Other CODECs, like G.729 and G.723, consume less bandwidth by performing compression, although doing so introduces delay and makes the voice quality very sensitive to lost packets. This does not mean you can't use G.729 across your wide area network — just make sure your WAN has little or no packet loss and minimal delay before considering a high-compression CODEC.

[1] - G.711 CODECs are supported by the Interaction Center Platform technology and associated Interactive Intelligence product suite. This does not guarantee, however, that your endpoint will support G.711. Please check with your hardware vendor to determine CODEC support.

Voice Compression (continued)

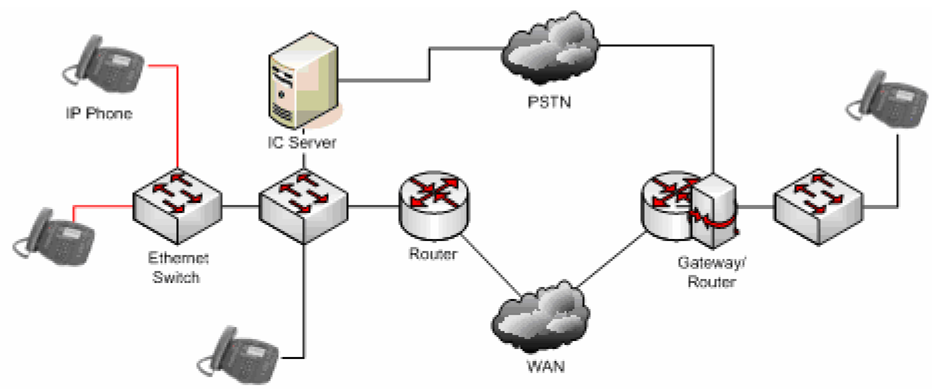
Deployment Tip: CODEC charts specify only unidirectional (one-way) audio.

Most charts discuss CODEC use in a unidirectional sense. The cost in bandwidth may double if your link is not full-duplex. Make sure you take this into account when sizing your network.

Also, the bandwidth consumed is usually more than the optimal data rate shown in Table 1. In most TCP/IP networks, for example, Ethernet, RTP and TCP/IP headers can increase the bandwidth requirements for a G.711 voice call from 64Kbps to 89Kbps. You can verify actual bandwidth usage with a VoIP calculator (www.voip-calculator.com). The Interactive Intelligence Global Services group or a certified ININ Partner can also help you verify actual bandwidth usage.

Finally, no matter which CODEC you use, it should be used consistently across your network. Moving from G.711 to G.729 across various points, for instance, will increase the number of Display System Protocols (DSPs) your network requires and will negatively affect voice communications quality.

Voice Quality Considerations



Paramount to the success of any VoIP deployment is the ability to maintain a high level of voice quality. More than any other factor, poor voice communications quality is what causes many VoIP rollouts to fail or come up short. But while some issues are unavoidable, the majority of call quality issues can be resolved with the proper network design and VoIP implementation. Following are three issues that most affect voice quality in VoIP networks.

Delay

Delay is the amount of time it takes a voice packet to be created, sent across the network, and converted back into sound. Given the necessity of encoding voice and the number of devices involved in a packet network, it is impossible to avoid some amount of delay in any VoIP network. The maximum acceptable delay for a voice network is 150ms.

> One-way delay = propagation delay + transport delay + packetization delay + jitter buffer delay

Voice quality drops quickly when the total one-way delay is greater than 150ms. In such cases, callers end up with a two-way radio effect in which they constantly "speak over each other." Here are the aspects of delay defined in more detail.

- <u>Propagation delay</u> is the time that voice communications require to travel the physical distance from end to end. For example, it may take a voice signal roughly 100ms to go from Dallas to Singapore. If your organization's voice traffic must cover long distances such as this, make sure your network path is as direct as possible.
- <u>Transport delay</u> is the total time a voice signal spends inside each device in the network, including switches, routers, gateways, traffic shapers, and firewalls. Some devices add more latency than others; for example, a software firewall running on a slow PC adds more delay than a dedicated hardware-based firewall. To determine transport delay times, look at the number of "hops" — the number of times a voice message traverses different nodes — that your voice traffic is forced to travel. Reducing the number of hops requires finding ways to reduce the latency in those devices that are the worst offenders.
- <u>Packetization delay</u> is the fixed time needed for the CODEC to do its job. Again, the G.711 CODEC imposes the smallest packetization delay. In contrast, CODECs that perform compression add delay, ranging from 25ms to 67ms. And remember, it's best to avoid converting from one CODEC to another along your organization's network path.

Deployment Tip: Actively minimize one-way delay, keeping it below 150ms.

Testing propagation and transport delay

One easy way to test propagation and transport delay is to use a TCP/IP ping to send a voice packet across the network and back. If you cannot perform a voice packet roundtrip below 250ms, your network will require optimization. Beware, however. Passing this test doesn't necessarily mean your network is ready for VoIP. As mentioned previously, ensuring a successful VoIP deployment requires a complete network assessment, so make sure whoever designs your VoIP network includes a well-designed delay plan.

Echo

Echo occurs throughout most voice networks today; as the delay in voice packet networks is higher, echo is much more noticeable. In effect, echo is caused by a variety of factors including wiring impedance mismatches and over-tuned gain on gateways as well as the Interaction Center Server (*Customer Interaction Center* or *Enterprise Interaction Center*) and various endpoints. Fortunately, many SIP phones, gateways and the Interaction Center Server provide echo-canceling buffers to remove some amount of echo from the audio stream. For more information, read the "SIP Application Note" from Interactive Intelligence. (See **References and Further Reading** at the end of this paper.)

Deployment Tip: Avoid low-cost analog endpoints at either end of the audio path.

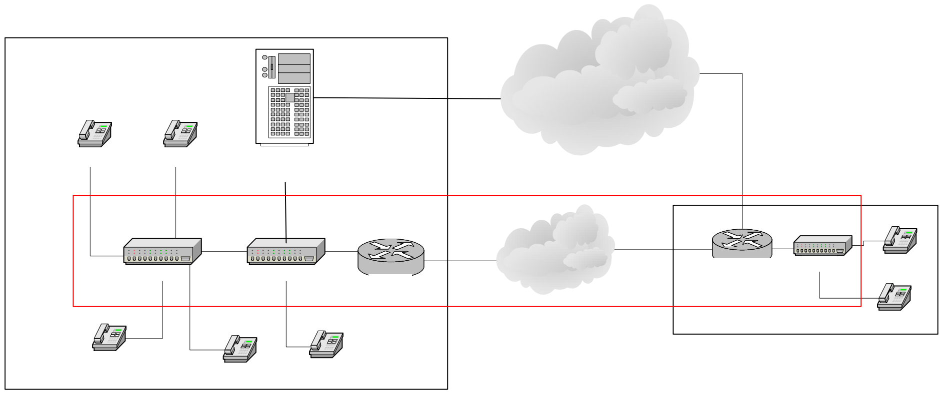
Another factor contributing to echo is inexpensive analog devices that aren't designed for SIP and VoIP. Avoid analog speakerphones attached through a gateway if possible, since these devices are notorious for causing echo problems.

Jitter

Jitter occurs when voice packets arrive at an interval greater than they are sent. For example, packets may leave the Interaction Center Server every 20ms yet arrive 25ms apart, which can cause obvious losses in voice quality since the voice communication will sound "stuttered." In cases like this, jitter buffers are used to dampen variations in packet arrival rates. Also if network delay is low and jitter is high, you can afford to have a larger jitter buffer than in a network where the delay is already high.

To get the most out of any SIP-based VoIP solution, organizations must achieve a balance between the jitter, delay and echo that varies greatly from network to network. Your organization can therefore expect a bit of fine-tuning with any VoIP deployment.

Quality of Service (QoS) Considerations



Quality of Service is a set of features that allow your organization to prioritize certain voice packets as they traverse your network. Without QoS, voice packets can suffer intolerable delays as they contend with other traffic (http, ftp, e-mail) that is much more tolerant of delay and packet loss.

The Interaction Center Platform event processing technology utilizes a standard known as Differentiated Services (DiffServ) to mark voice packets. Every packet leaving the Interaction Center Server (*Customer Interaction Center, Enterprise Interaction Center*) is set with the IP Precedence adjusted to 5, or Critical. This setting allows network administrators to differentiate voice traffic on switches and routers to ensure minimal latency.

Different vendors also will utilize this information for various queuing algorithms. Check with your network provider for QoS support and use the latest queuing techniques the provider recommends.

Deployment Tip: Use priority scheduling (QoS) for voice traffic.

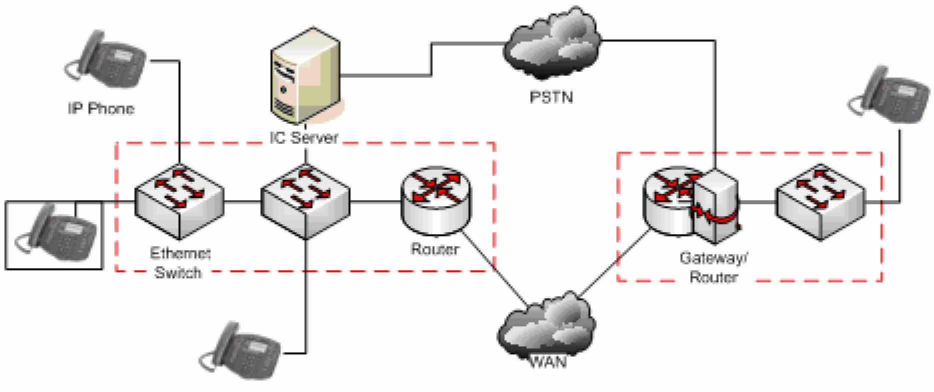
Without QoS in place, voice quality will suffer. To guarantee voice quality when using the IP solutions from Interactive Intelligence, QoS must be in place from every station to the Interaction Center Server. (Watch for adverse affects on your organization's existing data traffic, though.) QoS is the subject of numerous books and training courses. Contact your network provider for information specific to their implementation of QoS.

Deployment Tip: Keep packet loss well below 0.1% to avoid bursts of consecutive lost packets.

Minimizing voice packet loss is a large part of maintaining VoIP quality. Because each packet contains a subset of the overall communication, packets that are lost or dropped due to network congestion decrease the resultant voice quality.

Packet loss occurs because of congestion or electromagnetic noise. It can also occur when jitter is high and the jitter buffer is too small to compensate. Increased bandwidth and good tuning can often reduce network congestion, which in turn reduces jitter and packet loss. Also, VoIP packets are very sensitive to collisions. Avoid the use of hubs at any point in the network to avoid collisions.

Network Device Considerations



Selecting the network elements that make up your organization's VoIP/SIP infrastructure is a critical task. These devices must provide the capacity, redundancy, interfaces and feature sets required to successfully deploy a SIP-based VoIP solution.

The following items are worthy of noting when selecting network devices for a VoIP deployment. Network devices in the VoIP network should deliver:

Gateways

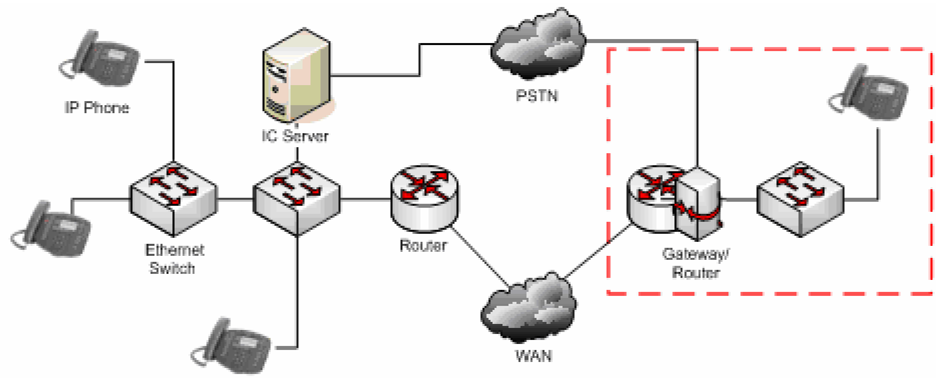
- SIP signaling (RFC 3261) support on TDM gateway devices
- Industry-standard CODEC support (G.711, G.729, etc.)

All Devices

- Wire-speed operation while under heavy load with Quality of Service (QoS) services enabled
- Hardware-based QoS mechanism supporting IEEE standard 802.1p/Q
- Provisioning tools for delivery of IntServ and DiffServ QoS methodologies
- Switching fabric capable of sustaining network bandwidth

Although the Interaction Center Platform technology supports enterprise and contact center applications riding over SIP-enabled VoIP, Interactive Intelligence continues to perform interoperability tests against common network infrastructure switches, VoIP gateways and endpoints. For an updated list of these tested devices refer to the "3rd party SIP Application Note" from Interactive Intelligence. (See **References and Further Reading** at the end of this paper.)

Remote Site Considerations



Many distributed organizations look to a VoIP network to link regional and local branch offices (and add new ones), to centralize call control between sites, and to avoid toll charges by sending calls over their existing data network. They also see a VoIP network as a vehicle for globally deploying data applications to remote offices — without having to install expensive equipment at each site.

Unfortunately, wide area networks can create new challenges and actually make existing problems worse. In addition to the network challenges they face, multi-site organizations must also consider those services that are critical to daily business functions at each remote location.

Local Calling

In most centralized VoIP deployments, all calls come in through a single entry point and are routed through that point (gateway) over the data network to the end station. With remote sites, however, you may not want all calls to be routed over a wide area network. As a WAN "bypass," organizations can instead attach a local gateway to the Public Switched Telephone Network (PSTN) for call routing — and avoid paying long distance charges at remote sites.

911

Along with local calling, users must have access to 911 facilities. Without a local presence (gateway), a site in St. Paul dialing 911 would be connected to the emergency facilities at your organization's central site in Chicago. At a minimum if replacing all stations at a remote site, consider a local gateway for 911 access.

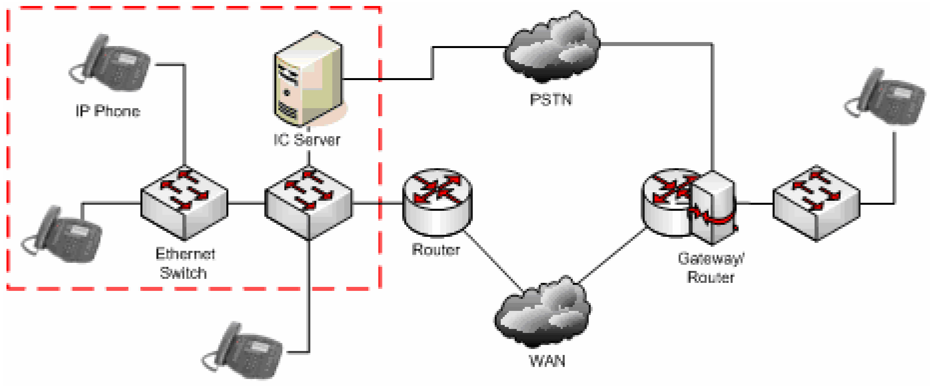
Dial Plan

When establishing a distributed network, dial plans can become more difficult to implement. For example, 555-1212 could terminate on any number of gateways depending on the area code from which it was dialed. To avoid unnecessary toll charges, make certain there is no room for confusion in your dial plan.

Voice Path

Finally, don't forget the voice path considerations referenced earlier: Be sure to account for the actual audio path through every network aggregation point. That is, know where your business is providing enhanced services — music on hold, recording, quality monitoring — and plan your bandwidth accordingly.

SIP-based Interaction Center Platform Functional Considerations



Before deploying VoIP over your data network, your own IT staff and system administrators <u>must</u> understand how the Interaction Center Platform technology utilizes SIP and how a voice call traverses the data network.

As an event processing engine, the Interaction Center Platform utilizes SIP to establish and tear down communication (voice) sessions. Advanced call control features like Conference and Transfer are managed at the Interaction Center Server level, allowing the Interaction Center Platform engine to utilize a variety of endpoints without proprietary additions to the SIP specification.

Basic Phone Call

Say Station 1000 wants to place a call to Station 1001. With the Interaction Center Platform processing engine, Station 1000 sends a SIP INVITE of sip:1001@ic_server. The Interaction Center Server will answer the call, play a prompt to the calling station, then send a SIP INVITE to Station 1001 (Figure 1).

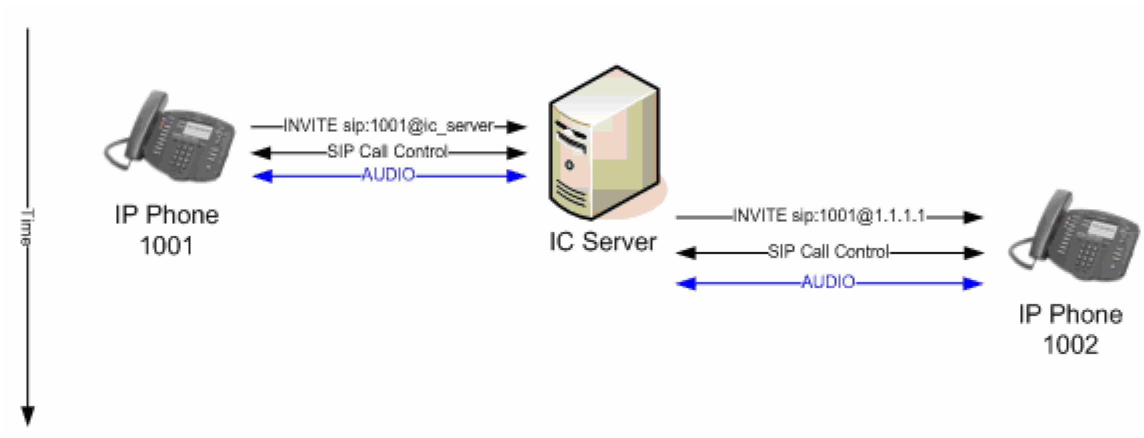


Figure 1: SIP Messaging for version 2.2 of the Interaction Center Platform (*Customer Interaction Center* and *Enterprise Interaction Center*)

This SIP Messaging process for phone calls frees the Interaction Center Server from having to rely on advanced SIP features that aren't yet standardized or found on all SIP endpoints. Moreover, the Interaction Center Server can play prompts to the caller, provide music on hold, forward calls, or utilize Find-Me/Follow-Me services with simple SIP phones and endpoints, such as Microsoft Messenger. All intelligence is maintained at the Interaction Center Server, allowing advanced functionality despite the relative youth of the SIP standard.

As shown back in Figure 1, two audio streams (one for each call leg) are generated with version 2.2 of the Interaction Center Platform technology. This additional bandwidth must be considered when sizing an Interaction Center-based VoIP network. The network will need to handle two audio streams from a local area network to the Interaction Center Server, even for a station-to-station call. As shown in Figure 2, this will not be the case in version 2.3 of the core Interaction Center Platform event processing engine.

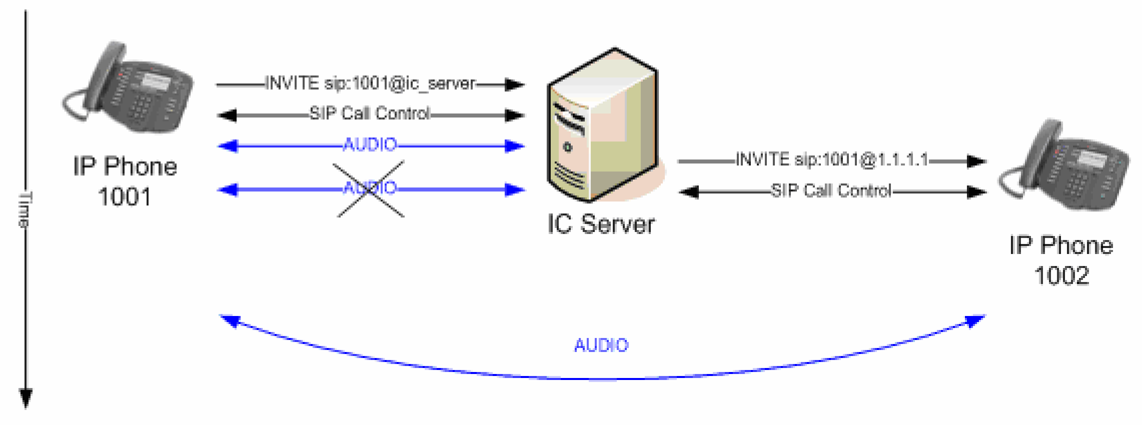


Figure 2: Audio Path for the Interaction Center Platform version 2.3 (and higher)

With version 2.3 and higher of the *Customer Interaction Center* and *Enterprise Interaction Center* software, system administrators can specify stations that will send audio streams directly to each other.

In this scenario, station 1000 will send an SIP INVITE of sip:1001@ic_server to the Interaction Center Server. The server will then answer the call and play audio (music, ringing) while it sends an INVITE to station 1001. Upon successful answer from station 1001, the Interaction Center Server will send a REINVITE to the phones, dropping the audio from station 1000 and establishing the audio directly from station 1000 to station 1001.

Interaction Center Platform – Advanced Functionality

Advanced media processing like music on hold, conferencing, listen, and record require the streams to be sent through the Interaction Center Server, regardless of version.

Listen and Record

The Interaction Center Platform provides other listen and record features by actively controlling the audio stream as discussed earlier. During listen and record sessions the audio flows through the Interaction Center Server, again regardless of which Interaction Center version you use (Figure 3).

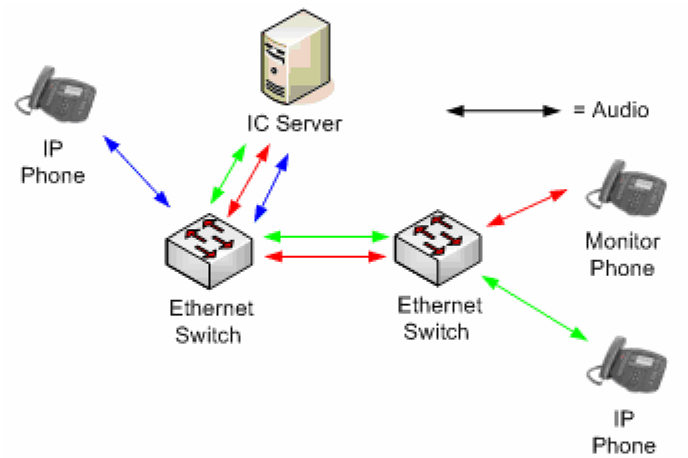


Figure 3: Active Record/Monitor Audio via the Interaction Center Platform

The Interaction Center Platform's listen and record process differs substantially from other VoIP solutions, where listen and record is performed passively on the LAN. Though it lowers overall bandwidth, the LAN-based method requires servers at each switching point to collect the audio stream (Figure 4). In some instances, supervisors can listen only to calls occurring on their own LAN. (For more on passive recording, ask your Interactive Intelligence sales representative or certified ININ Partner for information on ININ's TAPI recording solution.)

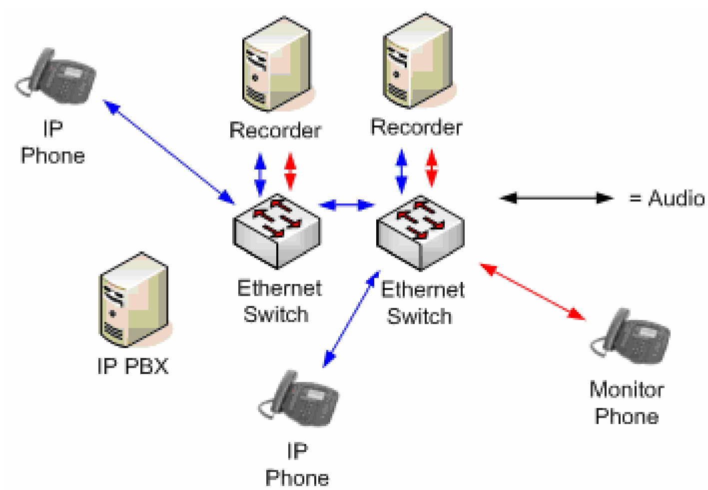


Figure 4: Passive Record/Monitor via a local area network (LAN)

Though active recording requires less hardware and simpler management, the additional audio will require additional bandwidth, which must be taken into account when designing a SIP-based VoIP network for the Interaction Center Platform.

Conference

Because the Interaction Center Server provides conference functionality, all stations go directly to the Interaction Center Server itself (Figure 5). Only one audio stream is required to each station, allowing up to 24 participants in a single conference. This differs greatly from many H.323 networks, where the audio is mixed either on the phone or using a hardware-based conference bridge (Figure 6).

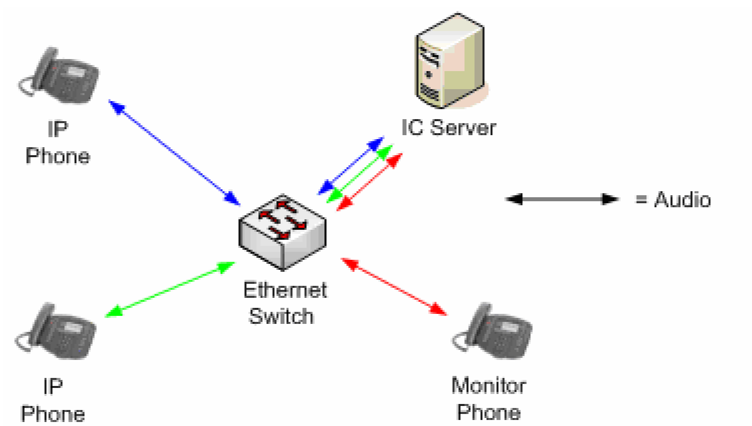


Figure 5: SIP Conference via the Interaction Center Platform

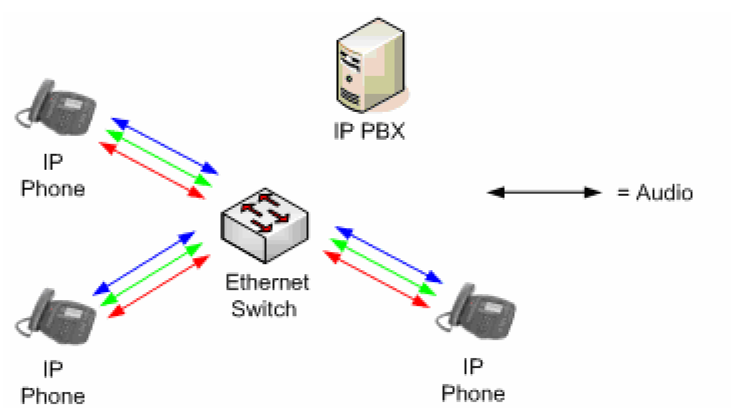
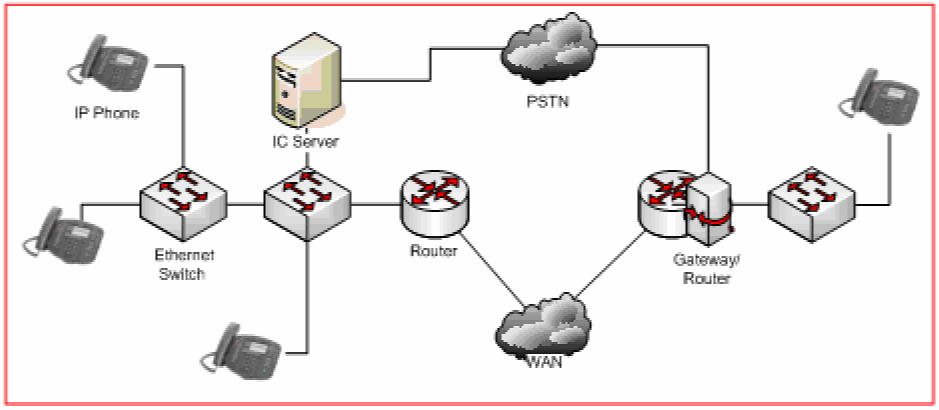


Figure 6: H.323-based Conference

For more information on SIP call flows via the Interaction Center Platform, refer to the "SIP Topology and Call Flows Application Note" from Interactive Intelligence. (See **References and Further Reading** at the end of this paper.)

Obtain a Network Assessment



Without question, the most critical step in deploying voice over IP is to complete a network assessment for every portion of your network tasked with carrying VoIP traffic. This assessment should be conducted by a VoIP engineer or technician experienced with the Interaction Center Platform technology and your particular network vendor. Your organization's VoIP assessment should include:

- Hardware and software inventories of every network device
- Detailed traffic analysis during normal and peak hours of network usage
- Review of current cable plant
- Detailed traffic planning based on peak call volumes
- Load testing of existing equipment

This assessment can be provided by Interactive Intelligence's Global Services group. Please contact your Interactive Intelligence sales representative or a certified ININ Partner for more information on ININ's Network Assessment services.

Deployment Tip: Get your data network ready for VoIP, fully upgraded and tuned, before starting a VoIP deployment.

In considering voice over IP for your business, the reality is that most data networks today aren't ready to carry good-quality voice conversations. *However,* it's easy to assess whether a network is capable of supporting VoIP, since VoIP traffic can be simulated so its characteristics can be measured and analyzed. Therefore, by simulating VoIP traffic your organization can make any and all changes needed in the network it uses, and can reasonably assure network success before launching an expensive VoIP deployment.

Appendix A: What is SIP?

The Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IEFT) standard for the setup and delivery of multi-media calls between two or more individuals connected to a TCP/IP network. The IETF RFP 3261 provides the framework in which SIP interoperates with other protocols (i.e., RTP, RSVP, YESSIR, RADIUS, and LDAP).

Within the SIP architecture are several key components worth noting. These components or SIP endpoints generally act either as a SIP User Agent Client (UAC) or SIP Server.

SIP UACs include:

- **Phone devices:** Phone devices are available both as standalone phones (hard phones) or softphones (software-based SIP-enabled clients). Make sure that any phone you purchase supports RFC 3261. For a list of phones tested for use with solutions developed on the Interaction Center Platform (*Customer Interaction Center* and *Enterprise Interaction Center*), see the "SIP 3rd Party Component Feature Matrix" from Interactive Intelligence.
- **Gateways:** These devices provide ingress and egress points from the SIP LAN/WAN infrastructure to the Public Switched Telephone Network (PSTN). Gateways are directly responsible for acting as "translators" for incoming/outgoing voice paths and their specific communications protocols, such as T1/PRI, SIP RTC, CODEC support, etc. Be sure that all voice/data gateways you use for SIP-enabled VoIP support RFC 3261. For more information, see the **Network Device Considerations** section earlier in this paper.

SIP Server devices include:

- **Proxy Servers:** These server components receive SIP signaling from SIP User Agent Client (UACs). A Proxy Server's primary job is to relay signaling requests to other proxies or UACs. They also provide services for authentication, authorization, network access control, routing, and security.
- **Redirect Servers:** These server components receive UAC requests and act as "forwarders," much like the analogous components in the Domain Name System (DNS) hierarchy.
- **Registrar Servers:** Typically these servers act in concert with SIP Proxies, allowing UACs to initiate SIP signaling with Registrars to update a database with their current locations in the TCP/IP Internetwork.

The Interaction Center Platform event processing technology provides both Proxy and Registrar Server services. For additional information on the need for SIP Proxy, Redirect and Location Servers, please refer to the Interactive Intelligence "SIP Application Note." (See **References and Further Reading** next page.)

References and Further Reading

Sources from Interactive Intelligence

Note: the Interactive Intelligence links listed here are located in the ININ website's password-protected Support Area. If you are not authorized to enter the Support Area, contact your ININ sales representative or a certified ININ Partner.

"SIP Application Note": (pdf format), March 2004.
http://www.inin.com/support/docs/AppNotes/SIP%20Application%20Note.pdf

"3rd Party Component Application Note": (pdf format) March 2004.
http://www.inin.com/support/docs/AppNotes/SIP%203rd%20Party%20Component%20Application%20Note.pdf

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Additional VoIP/SIP sources

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