GENESYS MEETING CENTER

VoIP

White Paper

June, 2006

Contents

Introduction

Genesys Conferencing is one of the world's first companies to offer a multimedia conferencing platform, which tightly integrates audio, web, multi-point video and enterprise solutions to provide the richest collaboration experience available in the marketplace today. With its commitment to innovation, Genesys is expanding its service offerings to include several variations of Voice over Internet Protocol (VoIP).[1]

VoIP is a technology that transmits voice over data networks. Its main application is to allow individuals to conduct voice conversations over an IP network instead of the existing public switched telephone network (PSTN[2]). This document will focus on VoIP solutions for enterprises.

The first VoIP application was presented in 1992, and, although the VoIP adoption rate has been slower than originally expected, it is now quickly gaining market share. In the short term, it is unlikely that VoIP will become a replacement for all components of traditional telephony. However, according to a report by Integrated Research, 78% of large companies worldwide are deploying IP telephony, in one form or another.

Drivers for VoIP Adoption

There are two major factors driving the adoption of VoIP within enterprises:

Lower Costs

- VoIP is a disruptive technology, helping to bring down the cost for telephony. Traditional suppliers, both in the telecom and network worlds, are now competing with a variety of new entrants, as Internet and IP connectivity can now be obtained from new players in the market.

- Similarly, we have seen new companies launching communications and PSTN solutions in the infrastructure arena. These new solutions or alternatives are offered at a fraction of the cost of traditional equipment, with more favorable and cost-effective service agreements.

- Enterprises will see improved cost-effectiveness by carrying voice and data traffic over one network, which will reduce operating costs. Many companies are merging data and telecom networks, which will help further reduce overall costs and drive the convergence of applications.

Increased Functionality

Increased functionality can also lead to enhanced productivity, resulting in cost savings:

- Enterprises will see increased business productivity through integration of voice functionality into business-critical IT applications such as customer service, company websites, "screen-pop" information on demand and "click to connect" capabilities.

- Presence awareness and "phone on the go" capabilities will become more prevalent. Wherever you log in and decide to make yourself available, you can reach out and be contacted via a single SIP address (superseding the telephone number) wherever you are: office, hotel or anywhere with IP connectivity.

- Unified messaging [3] and real-time collaboration will further spur increased enterprise productivity.

This paper describes the basics regarding VoIP, including its advantages and disadvantages, its deployment within enterprises and its impact on Genesys Meeting Center specifically. This paper also stresses the importance of investing in VoIP solutions that adhere to standards-based protocols, making it possible to reap the benefits of continuous competition.

In addition, the paper describes what Genesys is doing to serve customers who are starting to use VoIP. It explains how they can leverage their investment and maximize their return when using Genesys Meeting Center[4].

About VoIP

Components of a VoIP Network

An enterprise VoIP solution performs essentially the same function as your current enterprise telephone network. A typical configuration consists of an IP Private Branch Exchange (IP PBX)[5], an IP network, end user devices, and a VoIP gateway[6].

IP PBX

The market for IP telephony equipment is rapidly emerging and is anticipated to surpass $15 billion in sales by 2007, according to a recent IDC forecast. At the heart of this dynamic market is the IP PBX. It connects internal calls without routing them outside the enterprise facility. This way, all the users on the network can share a limited number of outside (trunk) lines for external calls. It can be significantly more economical and efficient than having a separate physical line for each phone on the premise.

Depending on the manufacturer and model, an IP PBX may perform other functions as well, including call transfer, voice mail, music on hold, and direct internal calling. In addition to performing the more traditional functions of a PBX, major IP PBX vendors can integrate with programs such as Microsoft Outlook and interface with other applications through an Application Programming Interface (API). This fact makes possible many of the new offerings that an IP PBX brings, such as unified messaging.

As enterprises move towards deploying VoIP as the standard for telecommunications, other business models will emerge. This will open up opportunities for service providers to offer a “virtual” PBX system where set-up, operations, hosting and maintenance are provided by third parties which service a large number of users, keeping service fees competitive.

An important factor to consider, regardless of the approach taken to implement VoIP, is to ensure that all components of a VoIP solution follow a standard such as SIP[7] (Session Initiation Protocol). Following standards will “future-proof” your investment, while not supporting industry standards will severely limit the use of new technologies, innovations, and alternate suppliers going forward.

IP Network

If the IP PBX is the heart of a VoIP telephone system, the IP-based data network is the blood vessels that carry the data. Again, this network serves the same function as the telephone wires in a PSTN. The same data equipment used for a corporate LAN should be up to the task.

End User Devices

Just as a PSTN phone is needed to carry out a conversation using the Public Switched Telephone Network, an IP-enabled phone is necessary to carry out a conversation using VoIP.

Softphones[8]

A softphone is software that enables telephone calls to be placed over an IP network. It runs on desktop computers, laptops, PDAs, or other endpoints. It has an on-screen interface that allows the user to make calls and utilize other telephone features.

High-quality headphones and microphones provide a considerable and measurable difference in call quality. If localized quality issues exist, spending money upgrading these two components will likely bring better results than upgrading the network or other software.

Hardphones[9]

A hardphone physically resembles a typical phone, with the ear-and- mouth handset, cradle, and keypad. The difference is that it has an RJ-45 Ethernet jack instead of an RJ-11 phone jack. This means the phone transmits and receives voice data packets over an IP network. Hardphones can use a variety of standard protocols such as DHCP[10] to configure themselves with the IP PBX.

Although a VoIP hardphone enables you to communicate over a data network, it still ties you to your desk like a traditional phone. However, a softphone provides users with mobility. You can be located and identified by your SIP address (described below), allowing you to take your laptop or PDA anywhere, promoting true presence[11].

Session Initiation Protocol (SIP)

SIP can be regarded as the enabler protocol for telephony and Voice over IP (VoIP). SIP is the standardized IETF (Internet Engineering Task Force) protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include VoIP calls and multimedia conferences. In addition, other text and multimedia sessions such as presence, instant messaging, video, online games and other services are enabled.

Users no longer need a phone number when using SIP. Instead, a SIP address will allow you to be contacted wherever you have Internet access. In the future, additional applications will use the same SIP address for features such as video and instant messaging.

An older protocol, H323,[12] will be sidelined or replaced, as it, with its PSTN heritage, has too many restrictions.

Purchasing and deploying an IP PBX or endpoint that is not SIP-standard compliant will severely limit your options by locking you into a particular vendor.

VoIP Gateway

A gateway is an interface that connects two different types of networks for the purpose of transmitting data. In a VoIP network, a VoIP gateway (sometimes known as a media gateway) is located at the edge of your VoIP network and acts as an interface between the IP network and the PSTN.

Multi-Protocol Label Switching (MPLS)

An MPLS-based VPN is the preferred method for accessing the Genesys Meeting Center because it guarantees an even, predictable, and quantifiable Quality of Service (QoS) across the network. MPLS uses labels on an IP packet to determine where to forward data and at what priority. Upon entering an MPLS network, a packet gets a label attached to it. This label contains data on how the packet should be routed. Once it arrives at the next router, the label is replaced with a new one, and onward the packet goes until it reaches its destination.

With explicit traffic engineering, you can direct packets to take underutilized paths, avoid congestion and other impediments, and generally balance network traffic. This feature captures the stability of the circuit-based PSTN without conceding the inherent advantages of a packet-switched network.

When aggregated, all these factors result in dramatically improved QoS. While stability and predictability are always desirable when sending and receiving data, they are imperative when it comes to VoIP calls. VoIP calls that use MPLS can enjoy significantly higher QoS than those that do not.

From a networking perspective, you will only need one connection to your service provider for your VPN, Internet access, and private link to Genesys. Thus, there will be minimal reconfiguration required on your router.

VoIP Advantages and Disadvantages

Advantages

Cost Savings

Cost savings is an often-cited driver behind the migration to VoIP. By utilizing packet-switched, instead of circuit-switched, technology, companies can leverage their existing IP infrastructure and use it as a data and voice network instead of having to maintain two separate entities (one for each function). At the same time, some of the inherent inefficiencies of the circuit-switched model, such as constant connections and the transmission of silences or dead air (when a caller is listening and not speaking) can be reduced or even eliminated. Making VoIP calls over a WAN or the Internet allows an organization to lower its toll costs for long distance calls.

Convergence

By piggybacking voice calls on existing IP infrastructure, it is much easier for organizations to implement a convergence strategy and to deploy truly integrated voice-and-data applications and unified messaging solutions. Features like initiating a phone call from your computer address book or receiving your voicemail in your email inbox are possible because of VoIP. The day will come when people will use one device connected to one network to access all their voice, text, and video communication, signifying true convergence.

Portability

Portability is another plus. By “untethering” users from the PSTN, incoming phone calls can follow you wherever you go. As long as you are connected to the Internet, you can be reached at the same phone number regardless of your location.

Increased Productivity

The above advantages contribute to increased productivity. By using the same network infrastructure to carry voice and data, you can streamline its administration. By integrating an IP phone into a unified messaging platform, you can reduce the clutter of communication methods people use. And, by making people more accessible, you lessen the time they spend trying to reach each other.

Drawbacks

VoIP has limitations, both technical and non-technical, that must be considered. Understanding how these limitations manifest themselves on your network and in your organization will help identify a solution that will make VoIP work for you.

Latency

Latency[13] is the amount of time it takes for a packet to travel from one endpoint to another. A real-time application such as VoIP places demands on an IP environment unlike those of other web applications. While you may not notice or care if an occasional email takes seconds to arrive, you will notice if there’s a similar gap in your conversation.

It is unlikely that a delay of that magnitude will disrupt your VoIP call, as latency is measured in milliseconds. Using the right codec (a scheme for compressing and decompressing a media stream), having enough bandwidth, and optimizing buffer settings can mitigate latency effects.

Jitter

Jitter[14] is a measure of the variability, over time, of latency across the network. Because of the nature of packet switched systems, packets may take different routes to reach their final destination. Even if they take the same path, network traffic and other variables may cause packets sent in regular intervals to arrive at different intervals. They may even arrive out of order.

A jitter buffer takes packets that arrive at different intervals and processes them into a regularly timed stream by turning the variable delays into constant delays. By regulating the buffer size, you can manage the trade-off between the amount of latency and jitter that users will experience.

Packet Loss

Packet loss [15] is when a packet becomes expired or misdirected and does not reach its destination. It happens to voice and other types of data packets, and it is an unavoidable fact of digital life.

While packet loss affects all applications, it is especially unforgiving in real-time applications like VoIP. Excessive loss can result in gaps in speech that, at best, will distort the sound quality and, at worst, the speech's intent.

The saving grace is that because of its prevalence, there are solutions in the marketplace that allow some of the lost packets to be recovered or, failing that, mask some of the effects of lost packets.

Other

Strategic considerations such as avoiding vendor "lock-in" in a fast-changing world and organizational challenges like changes in employees' roles and responsibilities are also factors that merit debate when deciding whether to pursue a VoIP solution.

Moving to Genesys and VoIP

VoIP may not be the right technology for every company, as every organization has its own requirements that need to be evaluated before deciding whether to pursue a VoIP strategy.

No matter what your decision is, Genesys Meeting Center can meet your conferencing needs. Genesys is completely agnostic and complementary to your existing choice of telephony equipment, whether it is PSTN- or VoIP-based, as Genesys has the ability to accommodate both types of customers.

Not ready to migrate to VoIP? You can still continue to use Genesys with your PSTN equipment and expect the same unparalleled meeting experience.

If you are interested in pursuing the VoIP option, teaming up with a mature and stable partner such as Genesys is a superior choice. With Genesys, there is no need to deploy your own IP PBX VoIP conferencing solution and assume the associated risks and costs. You would not need to devote any resources to its installation and maintenance. As Genesys is a conferencing specialist, our product is a full-fledged multimedia platform, accommodating features, such as application sharing, slide presentation, call management, video, and event management, that you may not have with your IP PBX.

In order to make all this possible, Genesys is following a blueprint that consists of three initiatives. The first one is aimed at creating a strong foundation for VoIP and the future evolution of our services. The other two are customer offerings completely integrated into our Genesys Meeting Center platform.

1. Backend Upgrade

Genesys is migrating from using traditional conferencing bridges to IP media servers. Media servers are carrier-grade equipment designed for multimedia applications, including conferencing. This migration improves Genesys reliability, scalability, and ability to innovate, which, along with the shifting realities of the IP telephony landscape, positions Genesys as the collaboration solution provider of choice for both IP and PSTN customers.

Media servers seamlessly support PSTN conferences, VoIP conferences and mixed conferences in which callers can be connecting from the PSTN, their mobile phone, an IP PBX and/or through a softphone.

This infrastructure investment also generates some very positive side effects. Though customers will see little change in how they use Genesys Meeting Center in day-to-day business communications, all customers stand to reap the benefits from the gains that Genesys makes in improving its infrastructure platform.

Better Failover

Dedicated connections, such as those used in PSTN, drop calls if there is a break in the connection. VoIP technology can simply reroute packets if there is a break on the network, with no disruption to the user.

Improved Scalability

Media servers have higher port densities than PSTN bridges. With the proper infrastructure and equipment in place, meetings with thousands of participants can occur.

Continued Product Evolution

Integrating media servers into the Genesys Meeting Center architecture allows Genesys to make product improvements that it could not in the past. Features such as the Genesys VoIP SoftPhone and mobile access make the meeting experience timely, fresh, and relevant.

2. Enterprise Connectivity

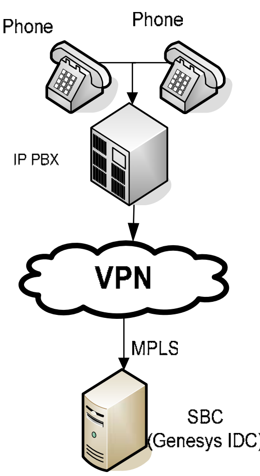
If your organization has invested in a VoIP network, you can capitalize on your outlay by connecting directly to the Genesys platform, without converting the signal to TDM.

Genesys Conferencing supports connecting to any SIP-compliant IP PBX via a VPN connection and to SIP-compliant devices, as well as most industry-standard codecs.

Connection via the public Internet is also possible. Transmissions over the public Internet can be encrypted; however, there is no QoS available.

At Genesys Conferencing, our research teams are continually designing and developing new services that will enrich tomorrow's virtual meeting experience. One such service Genesys is developing in response to our customer needs is to leverage additional IP PBX features, such as integrating with the IP PBX phone directory.

Recommended Enterprise Customer Connection Method



Genesys Conferencing is confident that our service brings you the industry's most comprehensive, advanced, and practical solutions for your virtual meetings. In response to customer inquiries, Genesys is focusing on leveraging the Application Programming Interface (API) for IP PBX vendors, starting with vendors that are SIP compliant, such as Cisco CallManager (CCM) version 5. Previous versions of the Cisco IP communications solution use SCCP (Skinny Client Control Protocol), a proprietary protocol owned by Cisco, to communicate with other components on its VoIP network, such as IP phones. Genesys Conferencing, like other major players in the VoIP arena, recognizes that standards, such as SIP, are the only way to realize the full potential of VoIP.

To implement a Genesys SIP and VPN-based solution will require an investigation of your current network. Genesys has a specific team dedicated to VoIP that will propose an appropriate solution for your business and assist in rolling out the service.

3. Genesys VoIP SoftPhone

In addition to delivering a SIP-based VoIP solution, Genesys has identified the need for a very simple but efficient solution that allows our customers to immediately reap some of the benefits of VoIP, while requiring very little investment and preparation.

We have developed a very efficient, low-bandwidth VoIP softphone that is shipped and integrated with our standard Genesys Meeting Center platform and is available to all of our customers. The Genesys VoIP SoftPhone is embedded in Genesys Meeting Center and unobtrusively co-exists with any existing softphone solutions you may already be using.

Current standards to effectively provide a reliable VoIP phone using the unreliable conditions common over the public Internet are still evolving. In the meantime, Genesys is deploying a proprietary softphone client that interfaces with our standards-based VoIP core network elements. As standards for public Internet IP telephony solidify, Genesys will ensure our softphone solution is compliant.

So, whether or not you have started to invest in VoIP, VPN or IP PBX, you can immediately start using the Genesys VoIP SoftPhone without conflicting with any long-term plans. Simply plug an industry-standard headset with a microphone into your computer, and you are “ready to go.”

Even though there is no QoS available over the public Internet, it is still the most practical method for using VoIP with Genesys Meeting Center. If you are traveling and use different Internet connections, the Genesys VoIP SoftPhone is an extremely convenient way for you to join the audio portion of a Genesys meeting. Whether you are going online from your home, a hotel, your office, an airport business center, a client’s office or a cybercafé, you can have one-click access to Genesys Meeting Center anywhere you have access to the Internet.

The first version of the Genesys VoIP SoftPhone, which is seamlessly integrated with Genesys Meeting Center, has several advantages:

Low Bandwidth

In contrast to many SIP (Session Initiation Protocol) phones that use as much as 80 kbps, you can attend the audio portion of a meeting using the Genesys VoIP SoftPhone with as little as 10 kbps of bandwidth.

High Latency Tolerance

Even with as little as 10 kbps of bandwidth and as much as 700 ms of jitter, users can still experience cell phone-quality audio even if the connection is less than perfect. In addition, the codec used is specifically designed for the conditions found on the public Internet.

Firewall Friendly

User-friendliness when negotiating firewalls is important, and the method by which Genesys bypasses firewalls is extremely flexible. UDP is the first protocol the Genesys VoIP SoftPhone tries to use because it is more efficient than HTTP, the default protocol for most web traffic. If UDP is not available, the Genesys VoIP SoftPhone will automatically fail over to another protocol. And, if that’s not available, it will automatically fail over to another one, and so on, until it reaches HTTP. The protocols tried in priority order are:

- UDP over port 4050
- TCP over port 4050
- HTTPS over port 443
- HTTP over port 80

In contrast, standard SIP phones often require manual configuration to open firewall ports and are thus less firewall friendly.

Additional Genesys VoIP SoftPhone Quick Facts

- Within an ongoing meeting it is possible for a participant to move between various audio access options such as streaming audio, the Genesys VoIP SoftPhone, SIP devices and PSTN.

- The Genesys VoIP SoftPhone will be embedded in the moderator install and can be centrally managed by IT and pushed to all desktops within an enterprise.

- Participants need to complete a small install to use the Genesys VoIP SoftPhone. This can be done prior to or within a meeting and does not require users to reboot their PCs.

- In addition to laptops, future versions of the Genesys VoIP SoftPhone will be available for mobile devices such as WiFi-enabled Smartphones and PDAs.

Minimum System Requirements

The following requirements are for moderators and participants using the Genesys VoIP SoftPhone within the Genesys Meeting Center.

Component	Requirement
Browser	Microsoft Internet Explorer 5.0
Pop-up Blockers	Permit pop-ups from *.iconf.net, *.conferencing.com, *.confarchives.com, and *.genesys.com
Java	Java and JavaScript must be enabled o Sun Java Virtual Machine version 1.4.2_06. (beta versions of Sun Java are not supported) OR o Microsoft Java Virtual Machine build 3810
Cookies	Session cookies must be enabled
CPU	400 MHz or higher
RAM	96 MB
Operating System	Windows 2000, XP
ActiveX	Required
Network Connection	56 kbps, of which 10 kbps will be used by the Genesys VoIP SoftPhone
Audio Hardware	▪ Stand-alone microphone and speakers, headset, or handset ▪ A USB audio device is preferable to one connected to a full-duplex sound card

VoIP Access

With the Genesys VoIP platform, there are numerous ways to use the Genesys Meeting Center. A meeting can occur with devices such as cellular, PSTN, or IP phones and conduits as diverse as the PSTN, Internet, or VPN. Different attendees can use different ways to participate in a meeting. The flexibility to suit your requirements is always there, whichever method you prefer.

Genesys IDC

Genesys web and audio servers are hosted by Tier 1 Internet Data Center (IDC) service providers. When you access the Genesys Meeting Center, your call, regardless of how it is placed, will be routed to a Genesys IDC. These IDCs host the bridges, media servers, web application servers, and other equipment that make your virtual meeting possible.

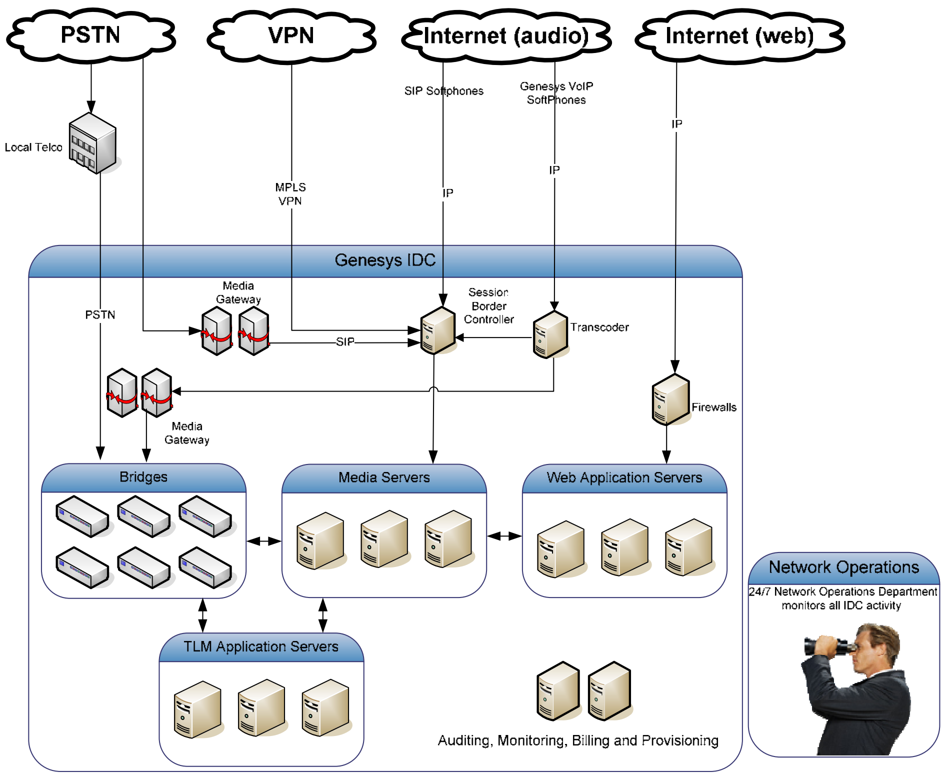


Figure 1 - Genesys IDC

Access Methods

Conduit / End User Device	PSTN	VPN (MPLS)	Internet
Phone (direct PSTN connection)	X		
Phone (w/ PBX)	X		
Phone (w/ IP PBX or Cisco Call Manager w/ SIP capabilities)	X	X	
Cellular Phone	X		
Genesys VoIP SoftPhone		X	X
Softphone with G.711 support	X	X	X
Browser			X

PSTN

PSTN is still the most common way to make a phone call. Genesys acknowledges its popularity by allowing the customer to use our services over the PSTN from most end-user devices. Local telephone companies carry these calls over the PSTN from your phone to a Genesys IDC.

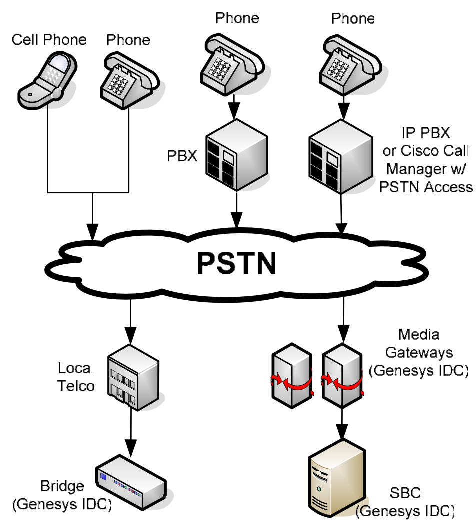


Figure 2 - Accessing Genesys Meeting Center over the PSTN

VPN

As stated earlier, an MPLS-based VPN is an excellent conduit for enterprises that have adopted VoIP to use the Genesys Meeting Center. It ensures an even, predictable, and quantifiable QoS. Most SIP-enabled devices can utilize this type of network connection to place calls to the Genesys IDC.

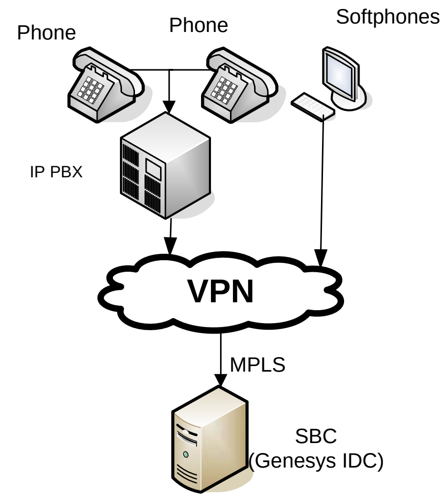


Figure 3 - Accessing Genesys Meeting Center over a VPN

Internet (audio)

If an MPLS-based VPN is not feasible or available, you can still use a wide variety of IP-enabled devices to connect to the Genesys Meeting Center through the public Internet. QoS is not necessarily guaranteed.

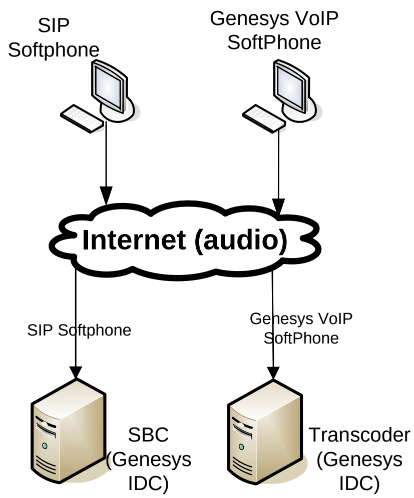


Figure 4 - Accessing Genesys Meeting Center audio features over the Internet

Internet (web)

With public, high-speed Internet increasingly accessible from more and more places, it is the perfect vehicle for using Genesys Meeting Center. Just turn on your computer and start your browser, and you can be in your Genesys meeting in seconds.

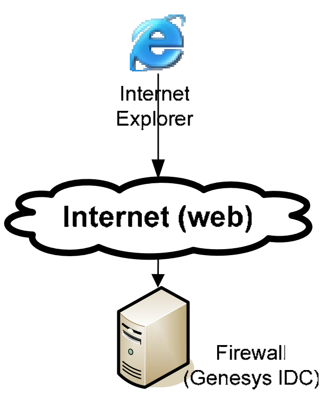


Figure 5 - Accessing Genesys Meeting Center web features over the Internet

VoIP Security

Genesys Conferencing employs many measures to ensure your meetings are secure. (See the Genesys Meeting Center Security White Paper for more details.) Genesys will apply the same diligence to ensuring that your VoIP meetings are equally secure.

Voice calls delivered over the Internet are susceptible to many of the same security threats as other web/Internet applications. With constant technological evolution in the field, lists claiming to be an exhaustive source of all the potential hazards will quickly become obsolete. Here are highlights of how Genesys is addressing some of the current potential security threats on its VoIP platform.

Potential Security Threat	Solution
Denial of Service (packet flooding)	• Network intrusion detection is part of the existing product security. • Session Border Controllers (SBCs) authenticate packets and prevent malicious ones from flooding the system.
Eavesdropping	• Using a VPN is an additional safeguard against eavesdropping.
Unauthorized Entry into Meeting	• PINs and passwords • Locking the Door – directs participants into a virtual waiting room until the meeting moderator is ready to admit them one-by-one or open the door and let everyone in. • Notification Tones – a double beep indicates to everyone in the meeting that someone has entered the conference. • Dismissing Participants – moderators can dismiss unwanted participants with a few mouse clicks or button pushes on the phone.
Physical Security	• Infrastructure hosted in Tier 1 IDCs

Contact Us

Thank you for your interest in Genesys Meeting Center. We would like to hear from you, and we're here to help if you have further queries.

Genesys Worldwide

Genesys Conferencing is a leading provider of integrated web, audio, and video conferencing services to thousands of organizations worldwide, including more than 200 of the Fortune Global 500. Our services are designed to meet the full range of communication needs within the large enterprise, from collaborative team meetings to high-profile online events. Genesys Conferencing's flagship product, Genesys Meeting Center, provides a single-platform multimedia conferencing solution that is easy to use and available on demand. With offices in more than 20 countries across North America, Europe and Asia Pacific, Genesys Conferencing offers an unmatched global presence and strong local support.

Additional information is available at www.genesys.com.

Key Terms - References

[1] VoIP is the means by which voice conversations are transmitted over packet networks such as the web, an intranet, LAN or WAN. Known also as Internet Telephony, VoIP allows you to unify messages by integrating VoIP-enabled voice signals with faxes and other data.

[2] PSTN is short for Public Switched Telephone Network, which refers to the international telephone system using both analog voice and traditional digital technologies such as ISDN.

[3] Unified messaging is the integration of several different communications media, such that users will be able to retrieve and send voice, fax, and email messages from a single interface, whether it is a wireline phone, wireless phone, PDA or Internet-enabled PC.

[4] Genesys Meeting Center allows you to hold interactive and productive virtual meetings anywhere at any time, with remote participants, directly from your desktop.

[5] IP PBX is a Private Branch eXchange (PBX) that supports the IP protocol. Typically deployed within the enterprise, the IP PBX is replacing traditional PSTN PBXs.

[6] A VoIP gateway is a network device that converts voice and fax calls, in real time, between the public switched telephone network (PSTN) and an IP network. The primary functions of a VoIP gateway include voice and fax compression/decompression, packetization, call routing, and control signaling.

[7] SIP is short for Session Initiation Protocol, an application-layer control protocol and a signaling protocol for Internet Telephony. SIP can establish sessions for features such as audio/videoconferencing, interactive gaming, and call forwarding to be deployed over IP networks, thus enabling service providers to integrate basic IP telephony services with web, email, and chat services.

[8] A softphone (from software telephone) is a program that enables VoIP telephone calls from laptop and desktop computers and other computing devices, such as PDAs.

[9] A VoIP hardphone is a self-contained broadband IP telephone that looks just like a regular phone. Instead of conventional phone jacks, VoIP phones have Ethernet ports through which they communicate.

[10] DHCP, Dynamic Host Control Protocol, is an IP-based protocol that allows computers and other IP devices to discover information about the IP network, such as IP address, and their default gateway.

[11] Presence conveys information regarding a user's availability and willingness to communicate. Used today in Instant Messaging, presence information will become prevalent in other communication media such as voice and video.

[12] H.323 is an umbrella recommendation from the ITU-T which defines the protocols to provide audio-visual communication sessions on any packet network. It is a part of the H.32x series of protocols which also addresses communications over ISDN, PSTN or SS7. H.323 is commonly used in Voice over IP (VoIP, Internet Telephony, or IP Telephony) and IP-based videoconferencing. H.323 was originally created to provide a mechanism for transporting multimedia applications over LANs, but it has evolved to address the needs of VoIP networks.

[13] This is the amount of time it takes a packet of data to move across a network connection.

[14] Jitter in VoIP terminology refers to the variability in packet delay which can have an impact on the quality of the voice conversation.

[15] Packet loss is the discarding of packets in a network when a router becomes temporarily overloaded or packets are corrupted or lost in the network.