Global Collaboration Services

VoIP Conferencing

The latest in IP technologies deliver the next level of service innovation for better meetings.

ENERGIZE YOUR CONNECTIONS™

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Executive Summary

PGi provides global business communications and technology-based solutions that enable PGi's enterprise customers to automate, simplify and improve communications with their stakeholders. One of the core solution sets in our portfolio is audio conferencing, which has become an indispensable tool in our daily business lives. PGi has always been a pioneer in this area, from the creation of the first reservationless audio conferencing in 1996, to the delivery of the first, native VoIP conferencing services in 2006.

This white paper will illustrate the critical advantages of our new VoIP Conferencing system by examining the following topics:

Traditional Audio Conferencing

This section describes the typical audio conferencing service widely deployed today so that we can see how the PGI technology addresses inherent issues in these systems.

What is VoIP.?

The term Voice over Internet Protocol (VoIP) has many different meanings depending on the context. This section describes how VoIP works in the context of audio conferencing.

A VoIP Conferencing System

The perceived simplicity of establishing an on-demand conference call is made possible by a multitude of machines and software performing a complex set of inter-related operations.

Core Customer Requirements

The user of an audio conference service has specific demands that must be satisfied for a quality user experience.

The Future of VoIP Conferencing

Insights into some of the new capabilities coming out of the R&D labs at PGI.

Traditional Audio Conferencing

Historically, the systems that provided audio conferencing services were a combination of the computing resources required to implement the service logic and the telephony resources required to provide the physical interface to the telephone network. These systems were usually called “bridges” in that they “bridged” together the audio conversations from all of the participants in the call.

Each bridge provided a fixed number of ports (aka participant connections) and there was no resource sharing between the individual bridges. The phone lines connecting each bridge to the telephone network were typically digital T-1 circuits and they were directly connected to the back of each bridge.

When a customer signs up for service, one or more conferences is created on a specific bridge. The customer is given phone numbers and passcodes to access their conference. Since a conference is assigned to a specific bridge, managing capacity on bridges is a challenge. Most conference reservations are “reservation-less” which allows the customer to dial into their conference at any time. The challenge for the service provider is to estimate the number of reservations that can be assigned to a bridge. If the bridge is under booked, then ports are not used. If the bridge is over booked, then customers may not be able to use their conference because the bridge is full.

In the event of a bridge hardware or software failure, it may be difficult to restore service to the effected participants. Since the T-1 circuits are connected directly to the bridge, re-routing phone numbers to new equipment can be a challenge. Typically, a bridge failure will result in loss of service for the customers who have been assigned conferences on that particular bridge.

To reduce risk of a bridge failure, bridges can be put behind a switch which can allow another bridge to take over in the event of a failure. Because of the architecture of the switch and bridges, the equipment is located in the same facility. If there is a site failure at the facility, it may not be possible to quickly and efficiently re-route the calls to another geographic facility which may cause a prolonged outage for customers.

With this overview of a traditional conferencing architecture and its limitations, the next section provides a basic understanding of VoIP before looking at VoIP conferencing.

What Is VoIP?

Until recently, all telecommunication services ran on the Public Switched Telephone Network (PSTN). This network was characterized by the design principle that says all intelligence must be located in the network core so that the telephone device can be extremely simple. This approach resulted in a service infrastructure that was incredibly reliable, but so inflexible and monolithic that new telephony services were very limited in what they could provide.

VoIP, which stands for Voice over Internet Protocol, is a new approach to delivering telecommunication services. Instead of a telecom specific protocol that requires a separate, dedicated network, VoIP uses the same protocols that are the foundation of the Internet. This approach means that phone calls are now just another form of data and can be carried on the same network as other Internet traffic.

Like much Web traffic, VoIP calls are considered sessions involving one or more participants. The Session Initiation Protocol (SIP) is an application-layer, human-readable request-response control protocol used to create, control and tear-down VoIP sessions. True to its Internet roots, SIP was created by the Internet Engineering Task Force (IETF) around the same time that the H.323 protocol was created by the International Telecommunication Union (ITU). SIP is transport independent and may travel over UDP, TCP or other transports. SIP messages are text based, and have many similarities to HTML.

Session negotiation and control through SIP enables the peer-to-peer nature of VoIP, removing intelligence from the network and putting it in the endpoint devices and applications involved in a VoIP session. SIP messages are addressed using the familiar Web construct of a Uniform Resource Identifier (URI). URIs allow VoIP endpoints to find one another using services which are already part of the Web, such as Domain Name System (DNS).

SIP messages typically carry a payload that describe the media codecs and attributes to be used in a VoIP call. It is the endpoint device or application that interprets the payload and establishes the corresponding media stream(s). Thus SIP enables, but does not dictate, negotiation of various audio and video resources and features delivered in a VoIP call. The audio and/or video streams travel over their own Real Time Protocol (RTP) sessions. These sessions are maintained by the endpoint devices producing or consuming them and are usually routed independently of the SIP messaging.

Throughout the life of a VoIP session, devices and applications may use SIP to renegotiate various aspects of the media stream(s), including the media codec(s) used and the endpoints of the various media streams. Typical uses include transferring a call, putting a call on hold, joining a participant to a conference, or changing the method of dual-tone multi-frequency (DTMF) detection. In the future, this could include dynamically changing the audio quality of a call through selection of a higher quality codec.

The symbiotic relationship between SIP as a call control protocol and RTP as a media streaming protocol provides a high degree of future-proofing for VoIP because SIP does not need to change in order to carry new types of payloads. Future media features, whether in the expansion of existing codecs, the addition of new codecs, or the creation of new streaming paradigms, are easily described and communicated in the current, flexible SIP payload format. In fact, the SIP payload need not be specific to media. It may be used to carry any type of information.

VoIP provides a fundamental change in the way that communications are carried. The next section applies this knowledge to VoIP conferencing.architecture and its limitations, the next section provides a basic understanding of VoIP before looking at VoIP conferencing.

A VoIP Conferencing System

A system is a functionally related group of elements designed to perform a particular function and satisfy a specific set of requirements. Our VoIP conferencing system is a collection of application servers, media servers and PGi developed software designed to provide simple, yet flexible audio conference services.

The basic unit of the system is the conference service site which collocates all of the hardware and software required to implement the following core functional areas:

> Network Border Access: Redundant SIP Proxy servers provide the entry point for all SIP calls coming into the conferencing system.

> Authentication and Authorization: The callers identity is verified at the Voice Service Director and authorization to use the conference service is provided at this point via communications with the Back Office Servers.

> Resource Load Balancing: A conference can run on any available resource within the system. It is the Back Office Servers that are responsible for managing the conferences spread across all of the available Conference Servers.

> Service Provisioning: Provisioning data is captured in a central Data Base of Record at a centralized site. This information is then replicated to each of our satellite sites. This approach means that the satellite sites are autonomous entities.

> Service Delivery: The delivery of the conferencing service is achieved through the coordinated efforts of two elements, the Conference Servers and the Media Server. The Conference Servers provide the business/application logic that defines the behavior of the service in terms of what prompts get played and what DTMF tones invoke specific conference features. The Media Server, under the control of the Conference Server is responsible for playing digitized audio prompts, capturing DTMF digits entered by the user and mixing the incoming audio streams from each of the participants.

The complete system consists of functionally identical conference service sites that are geographically distributed to provide service delivery during disaster events. Total load is managed so that it is possible for the system to suffer the loss of an entire site without loss of capacity. Conference provisioning is structured such that conferences can run at multiple sites. With this understanding of PGi's VoIP conferencing services, the next section provides the inherent benefits to the customer.

Core Customer Requirements

While previous sections focus on VoIP technology and how PGi uses it, the goal of this section is to highlight why VoIP is important to the customer – because it meets customer core requirements for reliability and quality.

The customer's definition of reliability is "anytime access from anywhere using any phone without any interruption to ongoing conference calls." There are an infinite number of ways that we could fail to meet the customer's expectation for reliability, but some of the common ones include hardware failure, underestimating capacity demands and scheduled service windows.

PGi's VoIP conferencing system architecture is designed to handle all of these conditions. Over the concern of hardware failure, the system provides for redundancy. As seen in the diagram above, calls are load balanced across the SIP proxy servers with any single server able to handle any call. Likewise, a SIP proxy server can call upon any application server to handle the call and any application server can call upon any media server across all the distributed sites.

The VoIP conferencing system allows for better capacity planning. Remember in the traditional conferencing environment that phone numbers are physically connected to the individual conferencing bridges. In VoIP conferencing, there appears to be a single "virtual" bridge that can be accessed by all phone numbers. The size of the "virtual" bridge and its shared resources eliminate the underestimating of conferencing capacity.

Scheduled service windows are designed not to affect customers in the VoIP conferencing environment. Traditional conferencing with its numbers and bridges tied together would be affected in a scheduled service window. When service was occurring on the bridge, the customer cannot access those phone numbers. Conversely, in a VoIP environment, resources are shared and while service is occurring on one server, the capacity of the other servers handles all the customer traffic.

The customer's definition of quality refers to the audio quality of the conference – can they understand all of the other participants and can those participants understand one another? Things that can hurt the conference's audio quality include echo, static, background noise, dropped audio and other artifacts. Most of the culprits are the same ones found in traditional conferences, although there are a few that are specific to VoIP.

Some service providers using VoIP to carry their calls will use CODECs that compress the audio signal to minimize the network bandwidth required for the call. This compression typically results in some loss of the audio information and the result is that the speaker tends to sound muffled. Dropped audio packets is another one of the culprits that is specific to VoIP. If a call traverses too many network hops or the hardware is insufficient to handle the load, then the network can elect to drop some packets to ease the load. If these dropped packets happen to be part of your spoken audio, then others may perceive that you said "ello" instead of "hello."

PGi's VoIP conferences match the audio quality of PSTN conferences by using the same 64 kbit CODECs used for traditional phone calls and by managing the physical network to engineer resources for the maximum anticipated load.

The Future of VoIP Conferencing

PGi is well recognized in the industry as one of the thought leaders for conferencing and collaboration services. As one of the few Conference Service Providers that develops all of their own service technology. PGi leads the industry with innovative new services that add value to audio conferencing.

> Native SIP Connectivity: The PSTN is still the predominate method of carrying phone calls and will remain for some time to come, but the time is swiftly approaching when VoIP will become the standard and replace the PSTN. The standards for VoIP are in an evolutionary state but become more defined also as time passes. These standards will allow conference calls from VoIP hardware and software phones completely bypassing the PSTN. With these improvements, VoIP would provide customers with an IP infrastructure that carries both voice and data.

> High Definition Audio: The PSTN provides a limited audio range up to 3kHz. Using the International Telecommunications Union's (ITU's) VoIP G.722 standard for speech CODECs, the range is extended to 7kHz, providing additional audio clarity. With the growth of international participants on conference calls, the availability of high definition audio will make international calls more productive.

> Regional Conferencing Hubs: The PGi VoIP conferencing system now covers much of the United States. With the deployment of additional VoIP conferencing systems in other regions of the world, PGi will be able to serve customers within their region. These regional systems when interconnected will give the customer "local" dialing access, reducing the need for International Free Phone (ITF) numbers. In addition, audio issues will be minimized with the reduction in audio links.

> Complete Resource Failover: This feature will provide the ultimate in reliability for conference calls. Conferences, once started, will be maintained throughout the call. In the event of a failure, the system will pass all of the call details from the non-working to a working resource. The feature will virtually eliminate dropped calls.

Summary

This white paper provides a look into PGi's leadership position in the development of audio conferencing services - from our initial creation of our Reservation-less Audio Conference service to the next generation VoIP Conferencing platform. Our continuing focus to automate, simplify and improve audio conferencing communications provides our customers with critical advantages:

- Proven foundation in traditional conferencing
- Focus on quality and service for VoIP conferencing
- Visionary leadership for the future enhancements

These fundamental advantages provide our customers the most reliable, robust and redundant conferencing in the industry, allowing flexibility of service and superior value.

For the latest information about PGi, visit our Web site at http://www.pgi.com.

Learn More

Contact us today or visit our website to view customer videos or product demonstrations and get the details on all of our solutions.

About us

Headquartered in Atlanta, Georgia (USA), and with presence in 24 countries, Premiere Global Services (NYSE: PGI) provides an established customer base of over 50,000 customers with on-demand communication technologies-based solutions that improve interaction and simplify business process overall.

The Terminus Building
3280 Peachtree Road NW
Suite 1000
Atlanta, GA 30305-2422
tel 404-262-8000
fax 404-262-8888
info@pgi.com

pgi.com

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