Web-Enabled Call Centers—A Progress Report

Howard G. Bernett, Denise M. Masi and Martin J. Fischer

Work is under way to fulfill the promise of Web/call-center integration. Both equipment and design tools are improving.

While ecommerce and online customer service continue to grow, many customers remain reluctant to complete a Web purchase without first talking to a live agent. Surveys indicate that between 25 and 75 percent of online shoppers abandon their shopping carts before completing a purchase, primarily because the website lacks real-time customer service to answer questions or resolve problems.

Thus, it's not surprising that a Purdue University survey of call center managers showed that the most significant initiative started in 2000 was website integration with their call center. The idea is to take advantage of agent pools who can assist Web customers, by installing a "Customer Service" button on a Web page that visitors can press to request assistance. Some studies have shown that corporations will spend more to upgrade their online customer-service capabilities than on any other information-technology effort in the next two years, with the bulk of that investment going into front-end call-center systems, back-end Customer Relationship Management (CRM) applications and website enhancements.

On-Line Customer Service Alternatives

On-line customer service is provided by call center agents who communicate with Web visitors via email, agent callback, text chat and/or Internet telephony. These tools, however, are not equally or ubiquitously available. Most Web-enabled call centers can handle email and provide callbacks, and the deployment of text chat is rapidly increasing. But Internet telephony is just beginning to appear in pilots and trials.

Emails, currently the most utilized Web-integration technology, are implemented by putting a "Send Questions" or "Contact Us" button on the Web page. When clicked, a form is displayed; it's filled out and sent back to the call center where it is answered either by an automated process or routed to an agent to answer it. Emails, however, are not a real-time interactive process, and response times can average between 10 and 50 hours depending on the industry segment.

In agent callback, a Web visitor fills out an online form requesting an agent to call back on a certain phone number and at a particular time. An agent then places a normal telephone call to the customer through the Public Switched Telephone Network (PSTN). This, however, requires that customers have a separate phone line to accept the callback while maintaining their Internet connection; otherwise, the customer must disconnect from the Web session and wait for the agent to call back without the benefit of concurrently viewing the website.

With text chat, the agent and caller exchange typed messages, which is very similar to a chat-room environment or instant messaging. When a Web visitor selects a "Chat" button, a form is typically displayed requesting pertinent information. The information, along with the chat request, is then sent to the call center where it is analyzed to determine the best agent to handle the request. A Java applet is sent to the visitor's browser to set up the chat window, or the chat can be performed in a pure HTML page.

Internet telephony requires that the caller have a multimedia PC that includes a speaker, microphone and an Internet phone client, such as Microsoft's NetMeeting, that accepts Voice over Internet Protocol (VOIP) calls. When a Web visitor clicks on a "Talk to Agent" button, a form is displayed requesting information about the call and about the caller's modem and software. The completed form is then assessed to properly set up the call and to select the best agent to handle the request.

With text chat and Internet telephony, an agent can actively push Web pages to the caller; with some products the caller also can push Web pages

Howard Bernett is a senior manager at Mitretek Systems (hbernett@mitretek.org); Denise Masi is a principal at Mitretek Systems (dmasi@mitretek.org); and Martin Fischer is a senior fellow at Mitretek Systems (mfischer@mitretek.org).

to the agent. This process of pushing pages, called "Web collaboration," is a key capability of providing on-line customer service.

Web-Enabling Technologies

There are two primary technology alternatives that a call center can implement for Web integration. The first enables a traditional call center that has circuit-switched-based systems to support on-line customer service. The second is the implementation of an all-IP call-center infrastructure.

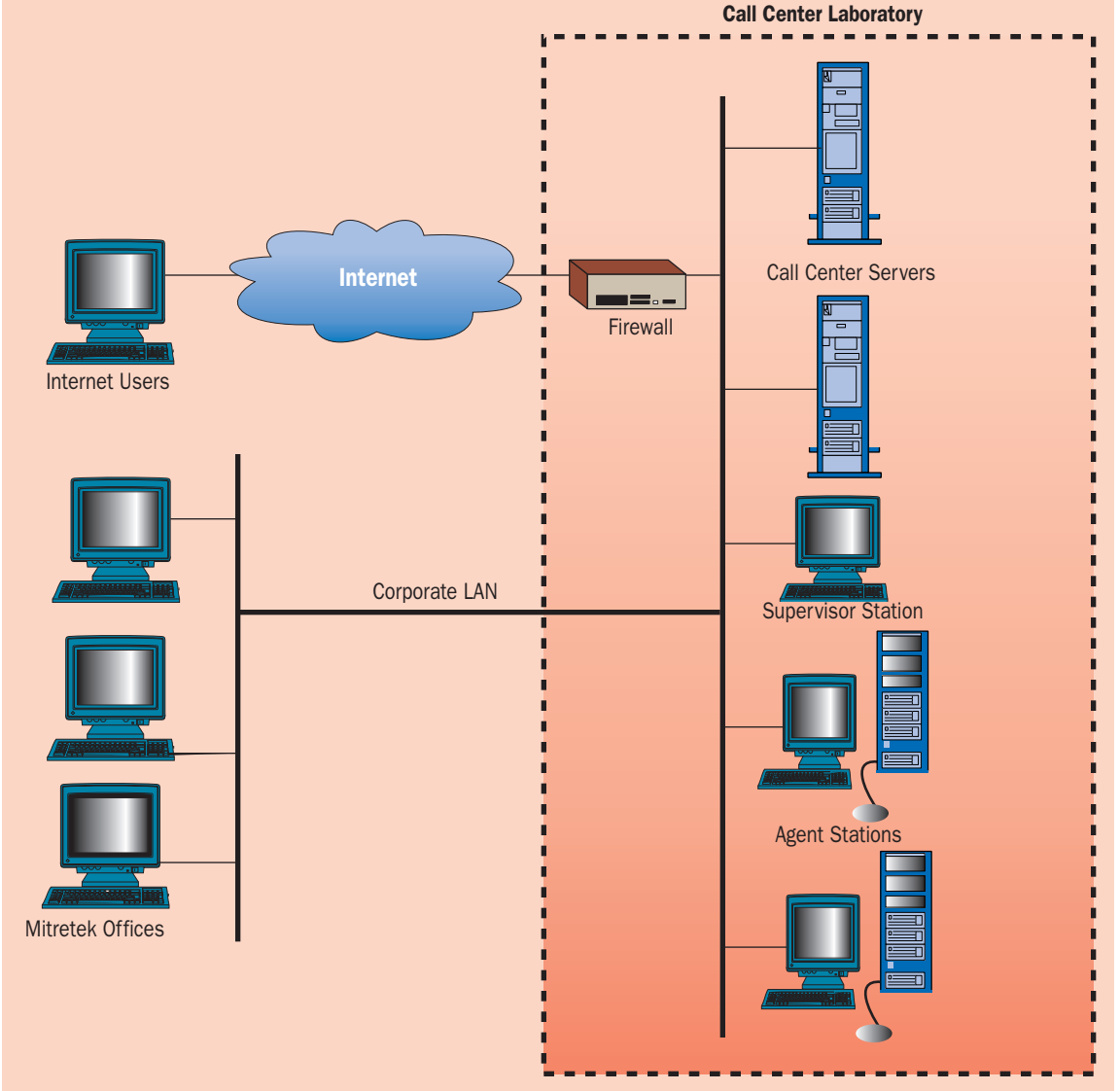
Most established call centers will enhance their existing traditional infrastructure to enable integration with the Web. This typically means providing bandwidth access to the Internet, installing an Internet call manager application, adding software to the existing ACD systems, CTI applications and agent stations, and connecting a VOIP gateway to the ACD. The Internet call manager provides the call-control function between the Web callers and the existing call center systems. The VOIP gateway converts VOIP calls from the Internet into time-division-multiplex (TDM) format and routes them to an ACD, where they are queued for an agent.

With an IP-based call center, the architecture is primarily a software solution that uses standards-based computer hardware, the TCP/IP protocol and WAN/LAN infrastructures. In this type of call center, an ACD server replaces the telephony-based ACD. The server performs the functions of a traditional ACD, but it does not physically switch calls; with IP, packets are addressed to the appropriate terminating device. The ACD server also provides universal queue control across all the contact channels and media types, and enables centralized administration and management reporting capabilities.

A call manager process is added, which establishes the connection to the caller and communicates with the ACD server to identify the first available agent that has the appropriate skills to

Mitretek's laboratory provides a neutral environment for evaluation

FIGURE 1 Mitretek Call Center Laboratory



The Call Center Laboratory has validated the superior upgradeability of IP systems

serve the customer. Based on the instructions from the ACD server, the call manager connects the call to the assigned agent. Since the call is not switched through the ACD server, the call manager creates the link between the caller and the agent.

To allow the call center to handle traditional phone calls, a PSTN gateway is utilized which converts TDM calls from the PSTN to an IP format. Since a PSTN caller does not go through the website, this gateway also performs the functions of an Interactive Voice Response (IVR) to request information from the caller. The IVR responses, as well as any other call-based information, such as DNIS and ANI, are passed to the ACD server to determine which agent should handle the call.

Improvements In Web-Enabling Technology

In mid-1999, Mitretek Systems, a not-for-profit public interest corporation, performed a market survey of Web-enabled and IP-based call-center technologies, which found that while a number of vendors had software in early stages of release, there were no enterprise-capable products or VOIP implementations for mid-size or large call centers. Based on the results of the survey, and as part of Mitretek's internal research and development program, we established a Call Center Laboratory to investigate and evaluate IP-based Web-enabled call-center technologies. The laboratory utilizes vendor-provided software running on Mitretek servers and desktop computers; Figure 1 is an illustration of the Call Center Laboratory infrastructure.

The primary mission of the laboratory is to evaluate the features and technology of Web-enabled products in order to determine the maturity of the offerings as they relate to the call-center market; the laboratory does not typically perform product-versus-product comparisons. The results of the evaluations are documented in published papers and live demonstrations are presented at conferences, such as Next Generation Networks and VoiceCon. The laboratory also provides a neutral environment where government and corporate call-center users can operate and assess the vendors' products without the pressure of a sales and marketing setting.

During 2000, the Call Center Laboratory received first-generation products that concurrently supported email, callback, chat, Web collaboration and VOIP. Although vendors were already shipping these products, there were very few actual cases of Web-enabled call centers. Indeed, a survey performed in 2000 by Utenberg Towbin found that only 10 percent of surveyed ecommerce websites offered text chat, only 1 percent had agent callback and none had VOIP.

Today, Web-enabled call-center deployments have significantly increased. In a Forrester Research survey of 50 call-center managers performed in 2001, 70 percent said that a Web-based call-center strategy was critical to their companies, and 26 percent had implemented Web call-center applications. To meet this demand, the vendors of traditional telephony-based products are providing Web-integration feature upgrades and delivering IP line- and trunk-side interfaces for their products.

Increasingly, new-generation, IP-based, call-center suites are becoming available from established vendors as well as from recently formed companies. These all-IP products include most of the features and functions found in traditional call center systems, and provide capabilities that were either very expensive, difficult to implement or unobtainable with traditional TDM-based products. Some of these include:

- The call center and agent locations can be independent—an agent can be placed anywhere in the world and be connected to the call center through the Internet.
- The agent desktop can support all media types — there is no need for a separate telephone set and computer.
- Straightforward integration with external and back-end applications (elimination of CTI) using Java, ActiveX, APIs, ODBC, etc.

As regards infrastructure issues, the IP-products introduce several significant benefits. Since the IP products are software loads installed on industry-standard servers, an all-IP call center does not require a traditional proprietary voice network with its associated PBX/ACD, punch-down blocks, point-to-point wiring and telephones. The elimination of this telephony network lowers the initial as well as incremental growth costs. For example, the Call Center Laboratory has been able to significantly upgrade its capabilities utilizing the servers and desktop computers that were initially purchased in 1999.

Another benefit is that a multi-site call center can directly connect the sites together through IP links without having to establish a mesh of tariff-based, circuit switched connections. This can considerably lower the annual WAN costs for the enterprise.

Challenges

While there's been notable progress, challenges remain, one of which is providing high-quality VOIP calls. Even though analysts predict that call centers may account for 30 percent of all VOIP products in the next two years, VOIP still has limitations with regard to quality of service (QOS) compared to PSTN calls. VOIP calls, when placed through the public Internet, suffer from problems related to packet jitter, delay and loss.

Most Web-enabled products that provide VOIP capability utilize the H.323 standard, and starting with Windows 98, all versions of Windows have included NetMeeting, which is an H.323 application. But while NetMeeting is currently the most prevalent call-center VOIP application, it does not

ensure a high QOS, and it requires the activation of several ports on firewalls that many corporations refuse to open because of security concerns.

Recently, VOIP products have been released that utilize the Session Initiation Protocol (SIP). Unlike H.323, which was originally developed for ISDN networks and then adapted for VOIP, SIP was designed to enable high-quality VOIP through the Internet. Many of the latest products also support the Media Gateway Control Protocol (MGCP), which interoperates with H.323 and SIP to create connections between media gateways. This enables an endpoint-to-endpoint connection through dissimilar networks.

Call center managers cannot design their operations LAN as if it were an administrative network and expect it to provide an acceptable QOS and level of reliability. The operations LAN must be engineered with sufficient robustness that it can survive failures that would cause a typical LAN to degrade or fail. This means that there must be sufficient reserve bandwidth available in case of router failure, and that the facility be designed with alternate cable routing between LAN rooms and core communications resources. In addition, routers should be able to prioritize packets and provide preference to voice packets.

Network managers also must ensure that inter-facility WAN links be procured through a vendor that can provide a service level agreement (SLA) that ensures acceptable VOIP packet performance. This means that the network must comply with ITU standard G.114, which states that a one-way delay budget of no more than 150 ms is acceptable for high voice quality.

There also are many challenges to implementing a call center that services both PSTN and IP calls—i.e., a "blended call center." A sufficient number of agents, adequate IP access bandwidth and other resources are essential to provide an acceptable level of service and to limit system congestion.

Queuing theory has been an essential tool for managers in estimating staffing and trunking needs in traditional PSTN call centers to meet performance goals. Blended call centers with both IP and PSTN calls, however, have not been mathematically analyzed. Performance of IP calls is a critical issue that call center managers have to control. Managers will find that IP calls and PSTN calls have different characteristics, thus complicating the allocation of system and human resources, for example:

- IP calls may have a longer call-handling time if Web collaboration is used.
- Arrival rates to the system of the two call types may differ depending on the popularity of Web use for the service or product.
- The bursty nature of IP packet arrivals to the call center, due to the alternating talk spurts and silence periods, results in aggregate packet arrivals with high variability.
- IP calls have an additional delay compared to PSTN calls, because of the buffering on the IP access line.
- IP calls do not require individual per-call trunks; shared-bandwidth access to the Internet is all that's needed.

We have developed a mathematical model of such a blended call center to understand its operation and allow call-center managers to provision resources. In particular, this methodology modifies basic queuing models to account for the different characteristics of IP and PSTN calls outlined above. Additionally, the IP access-line subsystem includes model components that account for the bursty arrivals of the IP call packets to this access line. We give the mathematical details for this work in the 2001 edition of Mitretek's *The Telecommunications Review* (www.mitretek.org/home.nsf/Publications/TelecommReview 2001; also see www.mitretek.org/callcenterscrm for all of Mitretek's call center and CRM papers).

Based on this methodology, we developed the Blended Call Center Analysis Tool (BCAT) in Microsoft Access Visual Basic for Applications, which is also described in the article referenced above. A network manager can use this tool to perform two types of analyses: First, to analyze the system performance for given design parameters; second, to design a system for a desired performance level. This tool can estimate the system performance by type of call, and then the impact of "what-if" scenarios can be seen.

The following scenario demonstrates how call center managers can determine blended call-center resource requirements using BCAT. For this example, the impact of decreasing the average talk time on system performance is investigated given a specific system design. The system parameters used in this scenario are:

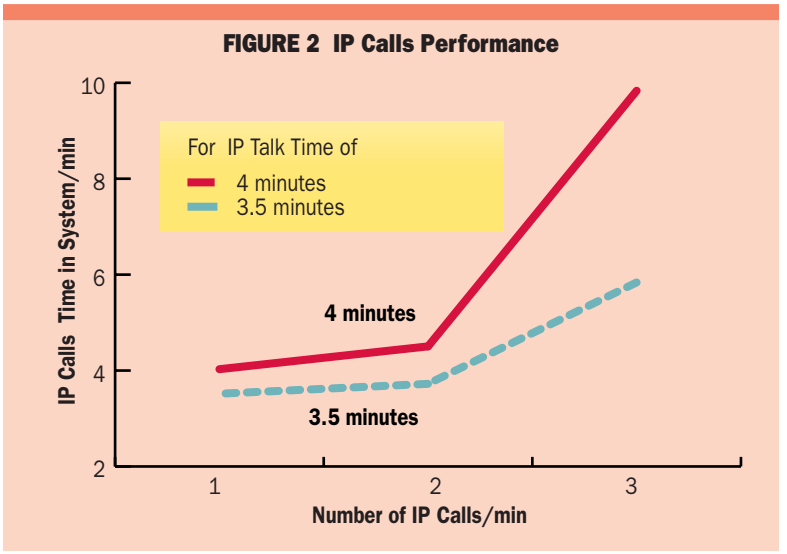
- Number of DS0 PSTN trunks = 30
- IP access line bandwidth = 1.544 Mbps
- Number of agents = 20
- Avg PSTN call arrival rate = 3 calls per minute
- Avg IP call arrival rate = Varies from 1 to 3 calls per minute
- PSTN call average talk time = 3 minutes
- IP call average talk time = Varies from 3.5 to 4 minutes

Figure 2 shows the performance implications of decreasing IP call average talk time from 4 minutes to 3.5 minutes. The time-in-system for IP calls is almost 10 minutes with an IP average talk time of 4 minutes, and a call-arrival rate of 3 per minute. When the IP average talk time decreases to 3.5 minutes, the in-system call time decreases to about 6 minutes (40 percent reduction of in-system time).

However, the analysis indicates that reducing talk time at lower traffic loads results in less of an impact on total in-system duration than at higher call volume. For example, the 40-percent reduction described above occurs at three calls per

We needed a new model for call queuing in a blended PSTN-IP call center

Over time, call center managers will move toward all-IP products



minute, a 17-percent reduction takes place at two calls per minute and 13 percent ensues at one call per minute.

Conclusion

The implementation of Web-enabled call centers is leaving the early-adopter stage and moving toward the mainstream. Web customers are beginning to expect that online customer service will be only a click away, so the companies that fulfill this expectation will take business away from those that do not.

We expect, however, that established call centers will continue to operate their traditional TDM-based equipment for many years to come. These call centers will implement Web-enabled and VOIP capability by upgrading their systems so that TDM and IP technologies coexist. Those responsible for new call center installations will have a choice—either installing traditional systems or the new all-IP products. Over the next several years, more call center managers will move toward the latter□