



More Features, Fewer Costs

Session Initiation Protocol (SIP) enables cutting-edge call centers to offer more applications while reducing daily expenditures.

By Mark Rayburn, Director of Advanced Technology at CPT International

Over the last few years, Session Initiation Protocol (SIP) has been making its way into call centers, telemessaging services, and customer service agencies at an increasing rate. Initially developed in 1999 as a signaling protocol for voice over the Internet (VoIP) conferencing, telephony, events notification, and instant messaging, SIP's popularity is growing. SIP's usage has expanded over the last five years, as entire networks are being built on this protocol. As usage grows, the quality of calls transmitted over SIP networks has improved immensely.

Call centers everywhere are starting to take advantage of SIP's guaranteed call quality and low costs. As uptake increases, they can begin taking advantage of the many applications, not to mention cost savings, which this protocol enables. Features like local number presence, remote service representatives, and better call routing to name a few.

Expanding Applications

Local-number presence is one SIP feature that call centers are beginning to capitalize on. For instance, a chain of pizza parlors that is overwhelmed by phone calls during rush hour may choose to utilize a call center in order to offload some of this heavy call volume. During these busy hours, all employees are making pizza, leaving no one to take phone orders.

SIP enables the pizza parlor to provide a local telephone number for placing orders, while still reaping the benefits of utilizing a call center halfway across the country. Customers simply call the local number in the phone book. In addition, the phone calls usually are answered in a more professional manner by a calm person in a relatively quiet call center instead of a harried cook in a noisy restaurant. From the caller's perspective, he feels as if he is getting great service: no repeating orders, being put on hold, or trying to be heard over clanging pots and pans.

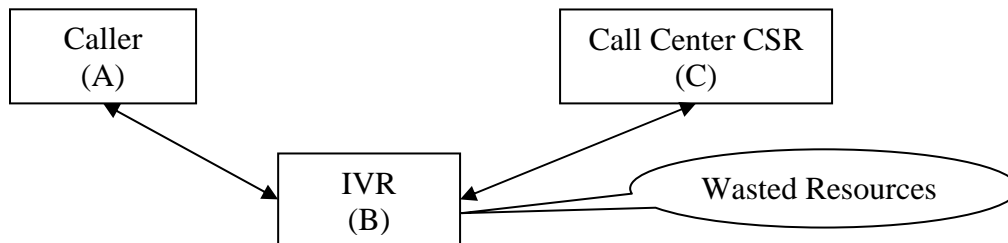
(Call centers can go the extra mile and provide the backend logic required to "remember" the last or most frequent orders. This is more efficient, which provides cost savings, and higher customer satisfaction and by feeling "connected" or making the service more personal. Also, take this opportunity to deepen branding and enrich the experience by piping in the aural wallpaper or ambience of choice like music, sounds, corporate jingle, etc.)

Local number presence is great for restaurants, but businesses of all types can exploit this feature. From local insurance offices and government agencies to dentist and doctor offices, businesses like the idea of offering a local phone number for customers to call even if the call is handled by a remote call center.

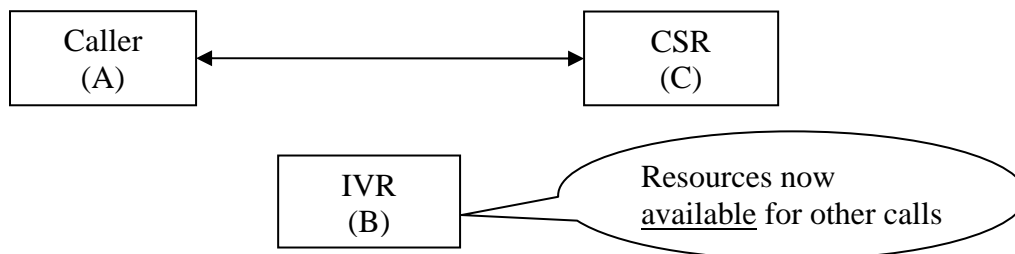
Local number presence is far more economical using SIP than using the TDM (Time Division Multiplexed telephone circuits, such as ISDN). In the TDM world, businesses are charged by the mile for a foreign exchange (from where the local number is to where the call actually terminates) or purchase tie lines.

Remote Agents: Another way cutting-edge call centers are taking advantage of SIP is by letting their employees telecommute to work. Rather than building an enormous office to house the call center and force employees to drive back and forth to work, call centers are providing employees with a broadband Internet connection, over which they can run both voice and data. Not only does this configuration save money on office space, but it also cuts down on the number of phone lines required for each employee.

Call Switching: Most call centers today sit behind automated call processing systems that handle the mundane, yet high volume calls allowing the humans in the call center to focus on more challenging tasks. These automated systems are typically referred to as Interactive Voice Response (IVR) systems. SIP provides IVR in economical and flexible ways to transfer calls to and from the call centers. Currently, call centers using TDM present challenges for the IVR front end when trying to perform advanced call routing call flows. For instance, a customer (A) calls the call center and connects to IVR system (B). Once the IVR system captures the caller's relevant information, it connects the call to a customer service representative (C).

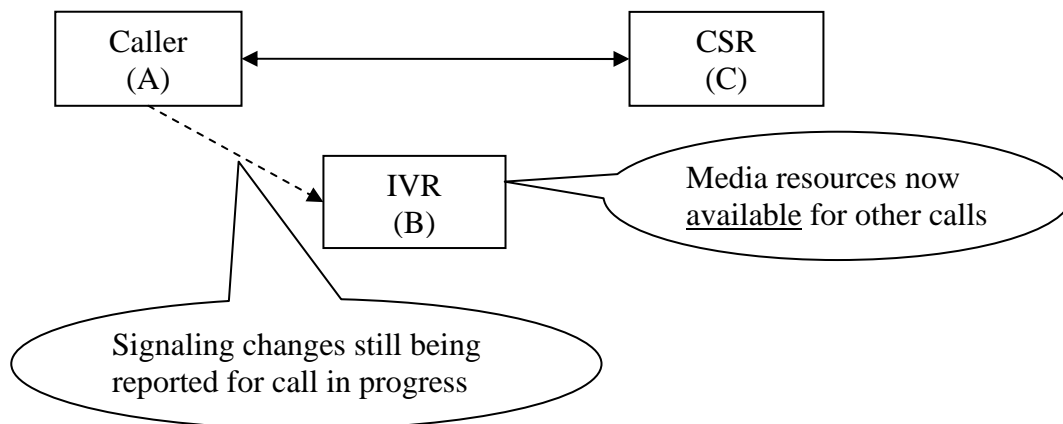


Although a two-way conversation is now taking place between A and C, B still needs to be in the middle to keep the call connected, possibly take back the call for further automation, and to track the disposition of the call (needed if billing is done from the IVR call records). This is wasteful of IVR, switching, and network resources. A more efficient call flow would drop the IVR (B) entirely out of the A:C conversation after the transfer is performed.



TNT (take back and transfer – an added cost network feature) could be used to connect A to C, but B would not know when either caller disconnected. Another challenge arises when the customer service representative wants to bring B back into the call, as there is no longer a connection. Depending on the carrier, additional provisioning and/or programming by the carrier can provide the CSR with a DTMF (touchtone) sequence they can enter on their phone to transfer the call back to the IVR (B). However, the IVR has to know that this is a continuing, rather than a new call, so a new DNIS (phone number) is required. Depending on how many transfer points occur in the application call flow, several new DNIS may be required. That just places the caller in the correct segment of the call flow. Now the IVR should not infuriate folks by asking for the previously obtained information again (such as account number, SSN, etc.) Methods for saving and retrieving IVR call context vary, but they all require a unique identifier that allows the proper context for a specific call to be fetched. With TNT, there is no easy way to pass this information back to the IVR. You are relegated to modifying the ANI (caller ID) or some out of band messaging over a data connection. It can be done, but it requires a lot of extra programming, and adds significant complexity. This design also lacks transportability due to the dependence on carrier cooperation and proprietary feature set.

SIP enabled call centers can offer features similar to Take Back and Transfer, but with even greater flexibility. One approach would be to employ SIP's "Subscribe and Notify" feature. In this example, A calls (SIP Invite) B, then B transfers (SIP Refer) the call to C. This effectively frees up B to take other calls and a conversation is connected between A and C. Optionally, if B (IVR) needs to track the disposition of the call, then B may issue a Subscribe and Notify. In this scenario, B still has freed up the media ports, but has retained a signaling path to the call.



When A and C hang up or press another touch-tone digit, Subscribe and Notify notifies B. This feature can't be accomplished in the current TDM world without embedded IVR in the carrier network, even with SS7 technology. Once B detected a signaling change, B could also redirect (SIP ReInvite) A to talk to B once again. This is just one high level example. There are many variations and schemes that provide many options for the SIP enabled call center.

Lastly, a SIP enabled call center can save operational costs by combining voice and data on the same circuit. No more separate vendor, equipment, and service staff for your voice circuits. Provisioning and scaling is dramatically simplified and faster to execute.

Implementing SIP

If you haven't already implemented SIP in your call center, you can more easily and affordably do so by outsourcing this part of your network and/or IVR system. In the past, if a business wanted to set up a VoIP connection to a local number, it would have to purchase a gateway, new routers, SIP-based phones, data circuits, and other expensive testing and monitoring equipment. However, an outsourced third-party hosting provider already has the equipment and services in place.

Additionally, a third-party hosting company greatly reduces interoperability problems; even though SIP is a standard, every equipment manufacturer has its own twist on this protocol, resulting in the potential for numerous issues. Third-party hosting companies already have worked through all the interoperability issues among the various equipment components. Seasoned MPLS based networks are in production to provide you a high quality of service. With the advent of IVR standards like VoiceXML, even the IVR can now be hosted safely with all the leading edge features like natural language speech recognition, speech synthesis, multi-language support, speaker verification, and even the beginnings of multi-modal services (the use of voice, touchtone, graphics, pen input, video, etc. – all on the same call using a wireless device).

SIP is an attractive technology to call centers because of its extensibility and cost-effectiveness. VoIP can save the call center significant costs, but the contribution value of SIP should be evaluated in terms of application features. Offer the customer higher satisfaction while realizing increased cost savings by engineering new functionality, not just TDM replacement functionality.

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[Author's Comment: This article refers to SIP specifically, but in fact, there are other VoIP protocols that offer similar functionality. The legacy H.323 protocol is still the most widely deployed, but I believe it is fair to say that the majority of new development is around SIP. However, H.323 is not static and v3 addresses many shortcomings of its predecessors; shrinking the differences between H.323 and SIP. I invite you to use your favorite engine to search for "SIP vs. H.323" and read all the latest comparisons. There are also some VoIP protocol newcomers, such as Skype, that continue to challenge the establishment and grow our options. So, for the more technically savvy, the term SIP in this article can usually be replaced with the VoIP protocol of choice and still hold true.]