Virtual Voice, Data and Video Solution Provider

SIP TRUNKING SOLUTION

Sever your ties to the old world phone network and make Digital Voice & Video (VoIP) calls directly from an on-premise IP-PBX or similar VoIP-enabled device or application. Why pay the monthly recurring charges associated with maintaining two distinct voice and data networks when you can realize significant and immediate savings by running voice as an application on your existing data network?

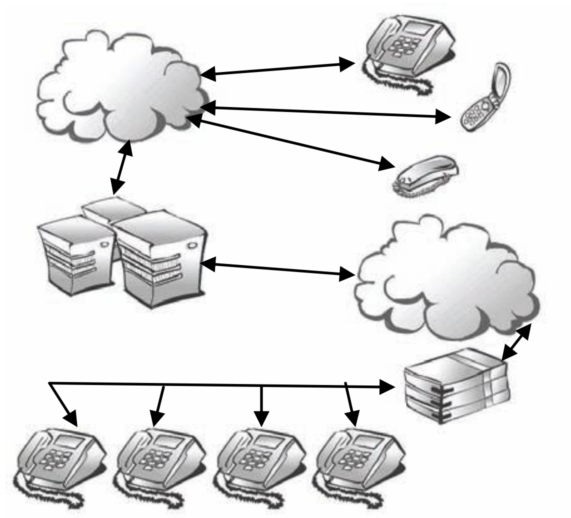
With the BlackCloud Networks SIP Trunking solution, calls are routed through the Internet rather than through the PSTN (Public Switched Telephone Network), providing your company with substantial savings by eliminating the need for costly T-1 PRI circuits and the expensive telco services that go along with them.

Keep your existing phone equipment. Keep your current phone numbers and/or add new numbers from over 10,000 cities in the US or 60 countries around the globe. Utilize data connections you already have installed and are paying for. Enjoy free calling between sites. For even greater savings and fully integrated Unified Communications (UC), consider our IP-PBX solution, but SIP Trunking does give the security and convenience of keeping your existing phone system and the features it provides. Think of it as an *evolutionary* step into VoIP as opposed to a *revolutionary* one.

BlackCloud achieves high call quality through innovative, patented technologies that allow us to monitor, analyze and react to the continual changes that occur not only in our own network elements but those of our supply partners as well. We analyze SIP signaling, node performance and IP metrics for all kinds of problems, ensuring that you receive the highest quality service.

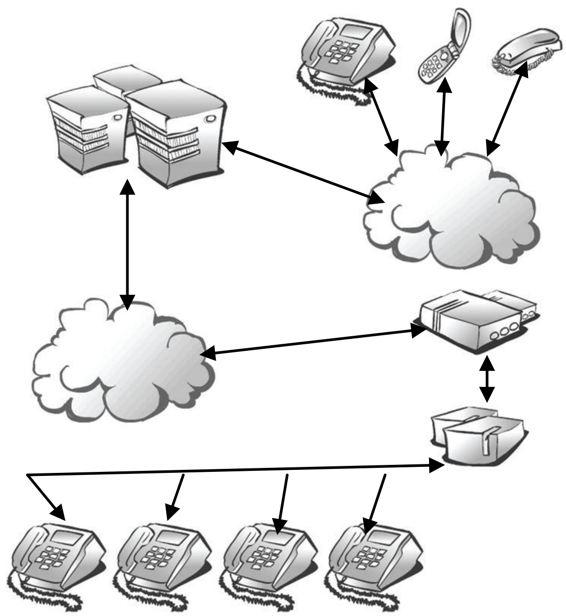
There are two flavors of SIP Trunking. The first—**Address Trunking**—is used if your existing phone system is already SIP enabled and can communicate directly with our SIP Trunking application. If you are using Allworx®, Shoretel® or any Asterisk®-based IP-PBX, we can provide you the phone numbers, 911 service and call termination you need.

How SIP Address Trunking works



The second—**Device Trunking**—is used if your existing phone system is not yet SIP-aware. A multi-port FXS gateway is installed between your phone system and your LAN. This allows your existing phone system to make and receive calls over the Internet. Because our application can address each port or line of your phone system through that gateway, we can offer you features like Trunk Groups, overflow voicemail to email and Caller ID management that your existing phone system must provide under Address Trunking.

How SIP Device Trunking Works



SIP Trunking Features

- Device Trunking Only: Web-based Administrator Control Panel
- (ACP) to manage all features from a single UI
- Device Trunking Only: Trunk Groups to optimize call flow
- Device Trunking Only: Overflow calls are sent to voicemail, then delivered via email
- Enhanced Caller-ID (Caller Name)
- Phone numbers from over 10,000 US cities plus over 60 Countries
- Toll-free numbers (ported or new)
- Full 911 coverage in US and Canada
- Inbound faxes delivered to email inbox and outbound faxing from most Windows applications
- Flexible service plans to meet your needs and your budget
- Add lines when you want—pay only for what you need
- Rapid service setup and activation
- Very low activation fees