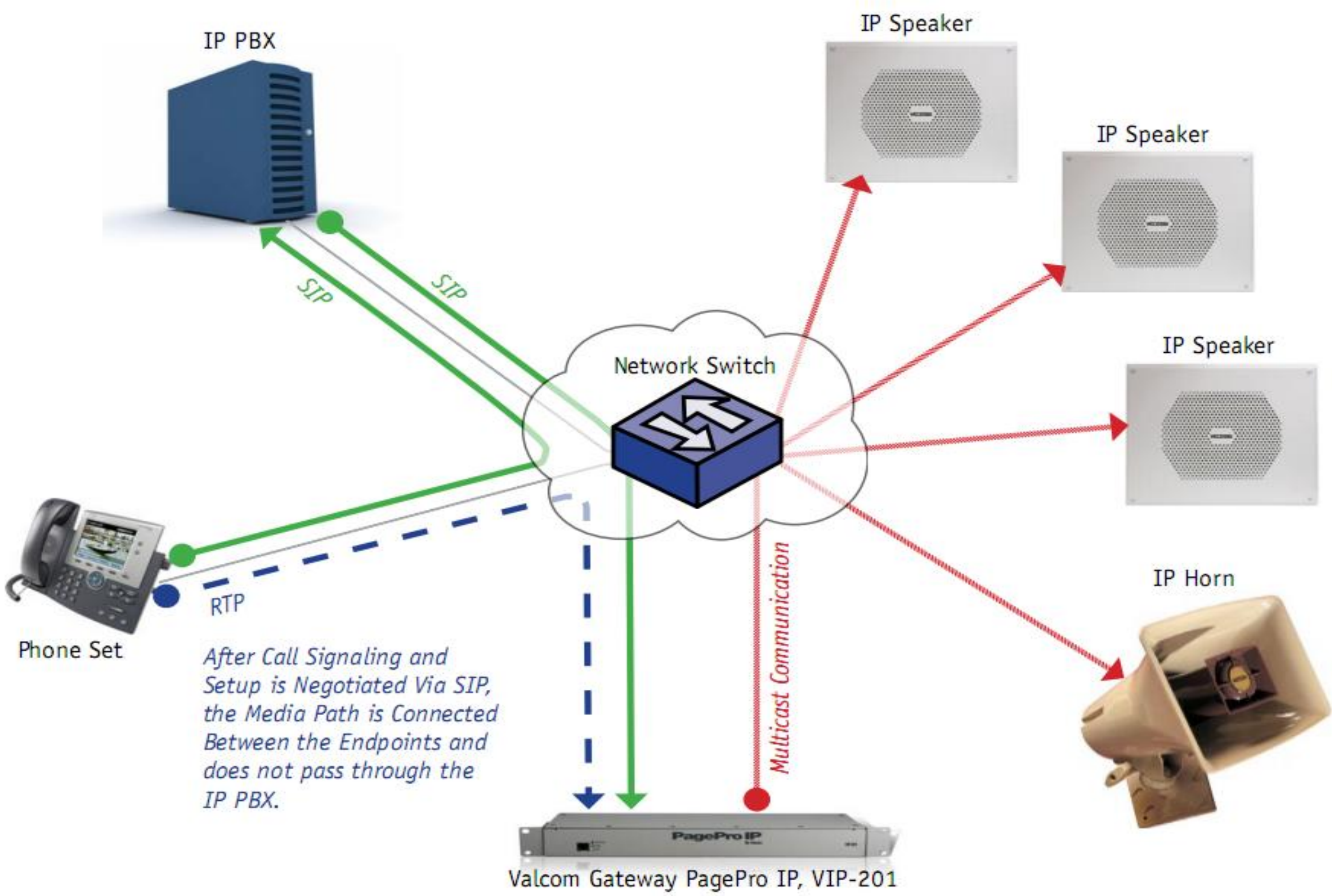
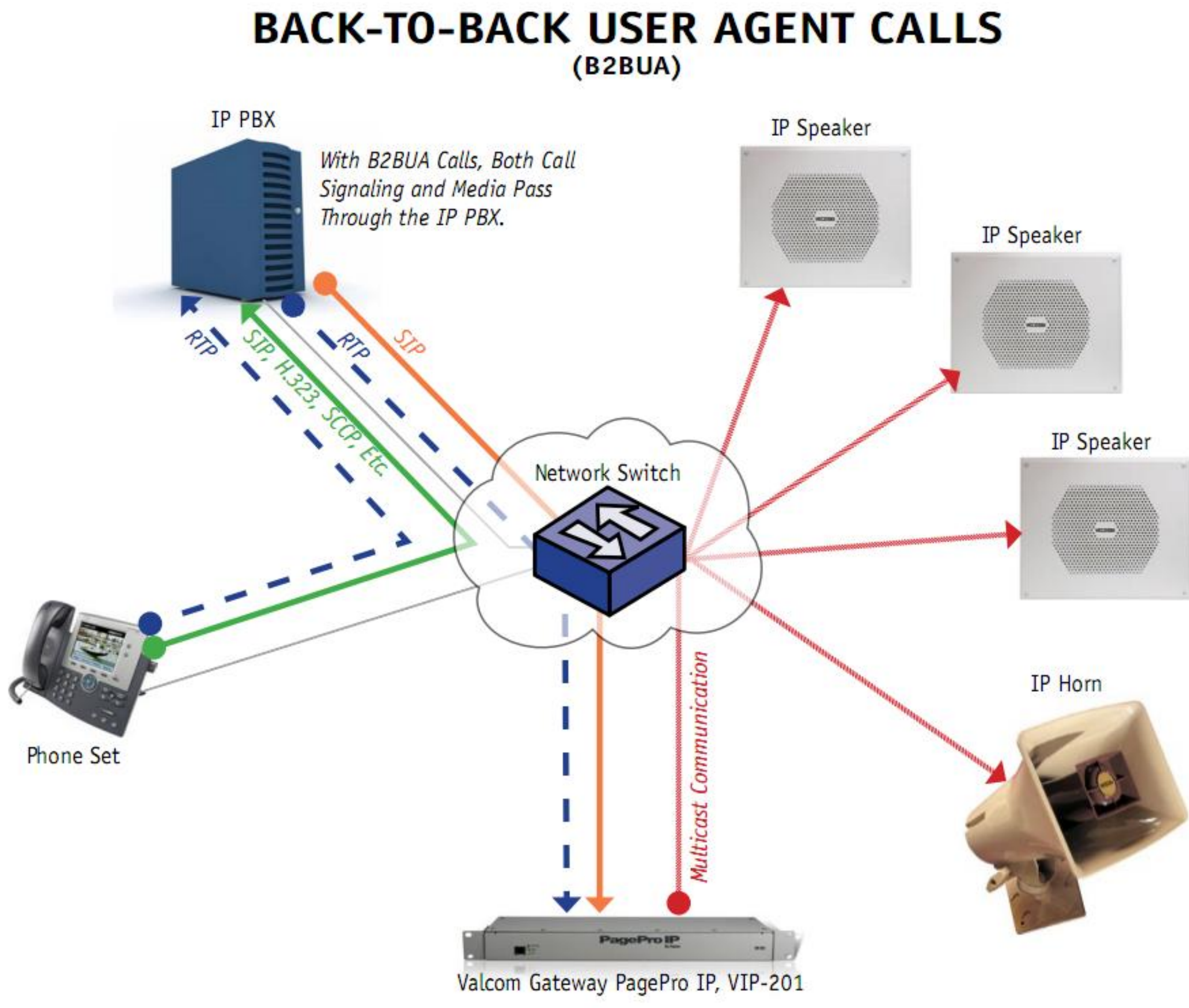
Session Initiation Protocol (SIP) Connection Options

Valcom SIP-enabled devices can connect to other SIP devices, such as IP PBXs, in several ways. The following diagrams illustrate the main connectivity options of the Valcom PagePro SIP Paging Server. Other Valcom SIP-enabled devices will connect in a similar manner, with varying capabilities depending on the device model.

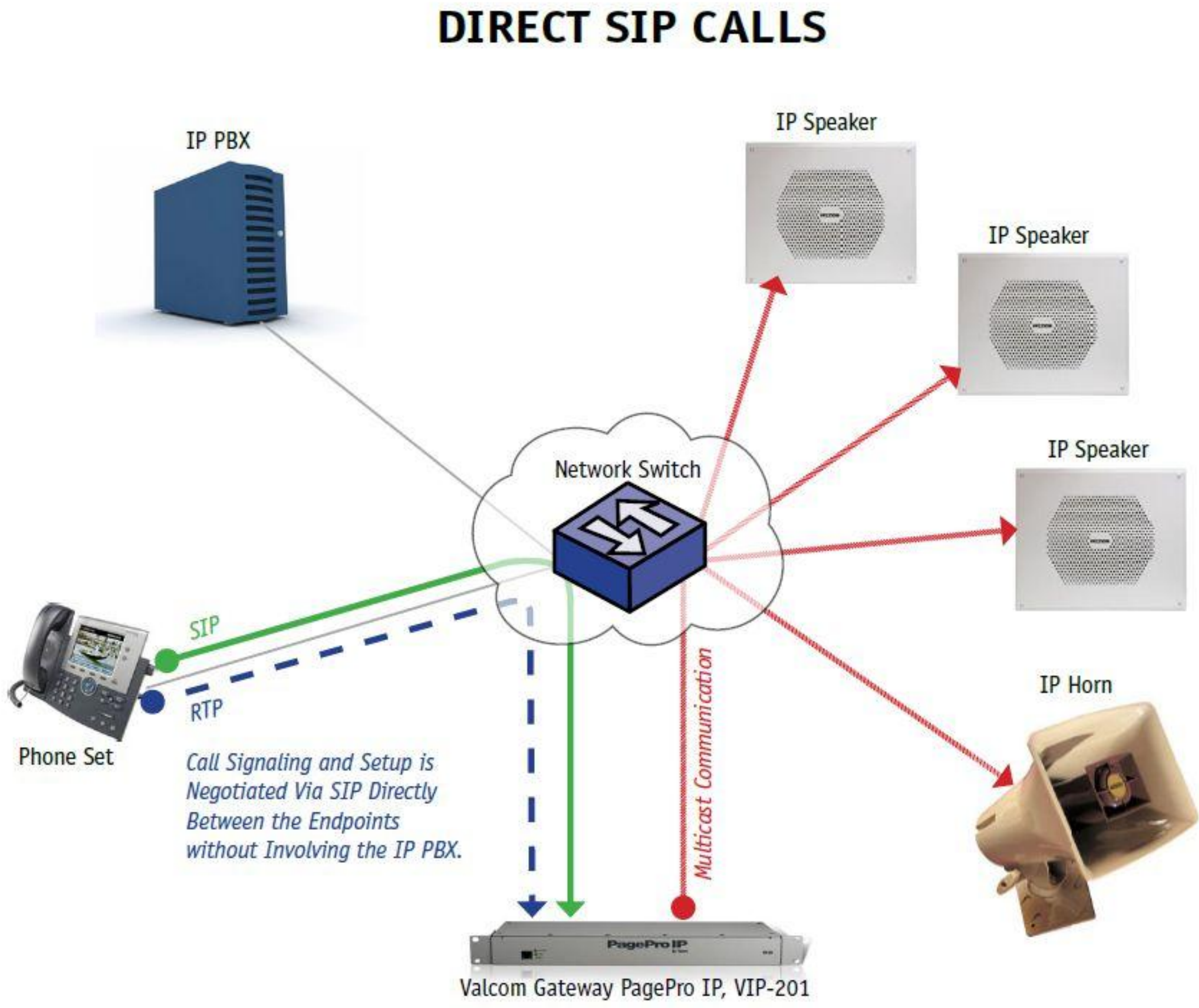
DIRECT MEDIA CALLS



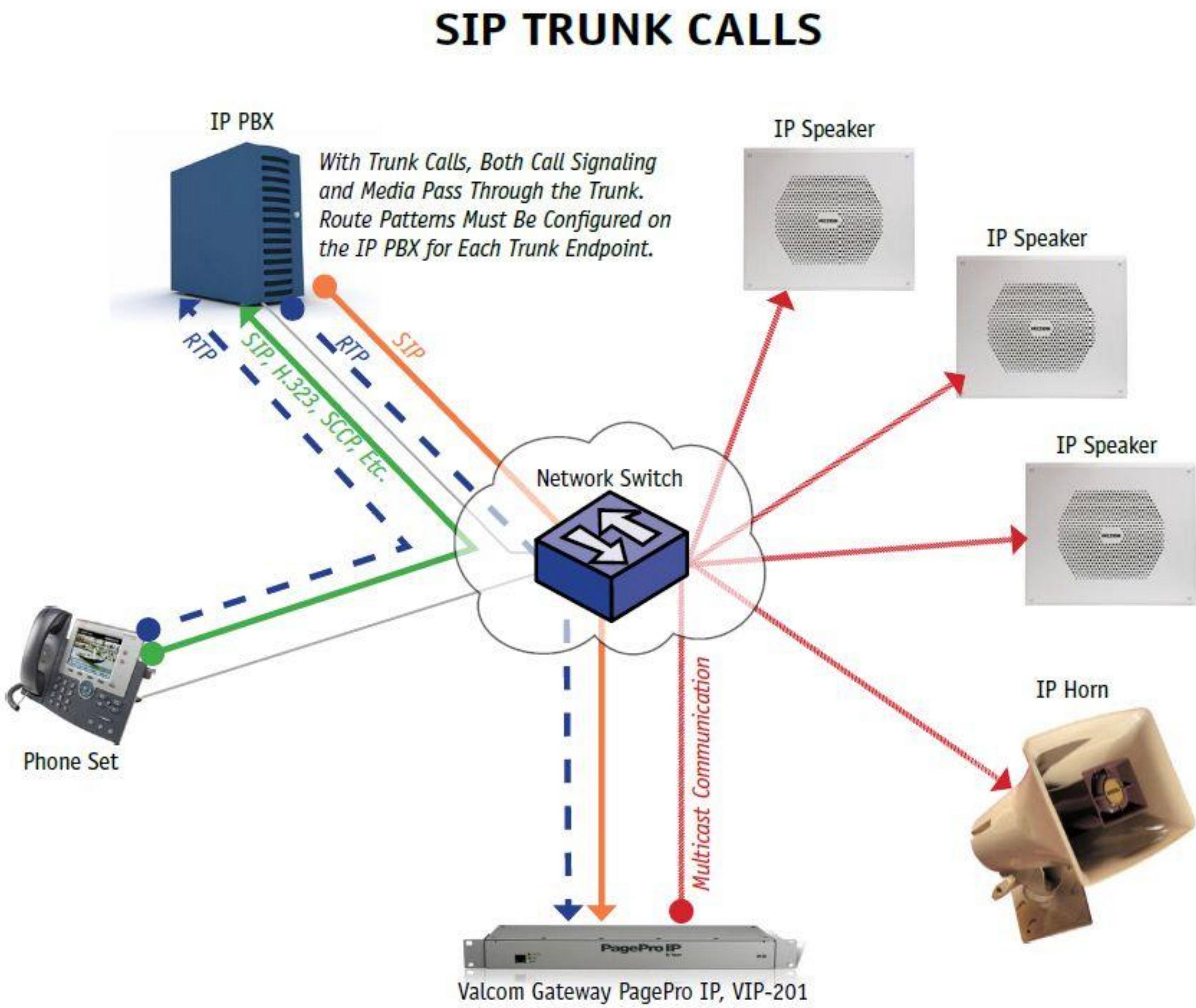
A telephone call with direct media allows the RTP data to bypass the SIP server, and flow directly between the endpoints involved in the call. When the call is completed, the endpoints send SIP messages to tell the SIP server the call has ended.



Back-to-back User Agent (B2BUA) calls keep the RTP data flowing through the SIP server. This is commonly used when information about the call needs to be maintained by the server, such as for billing records or to record the audio of the call.

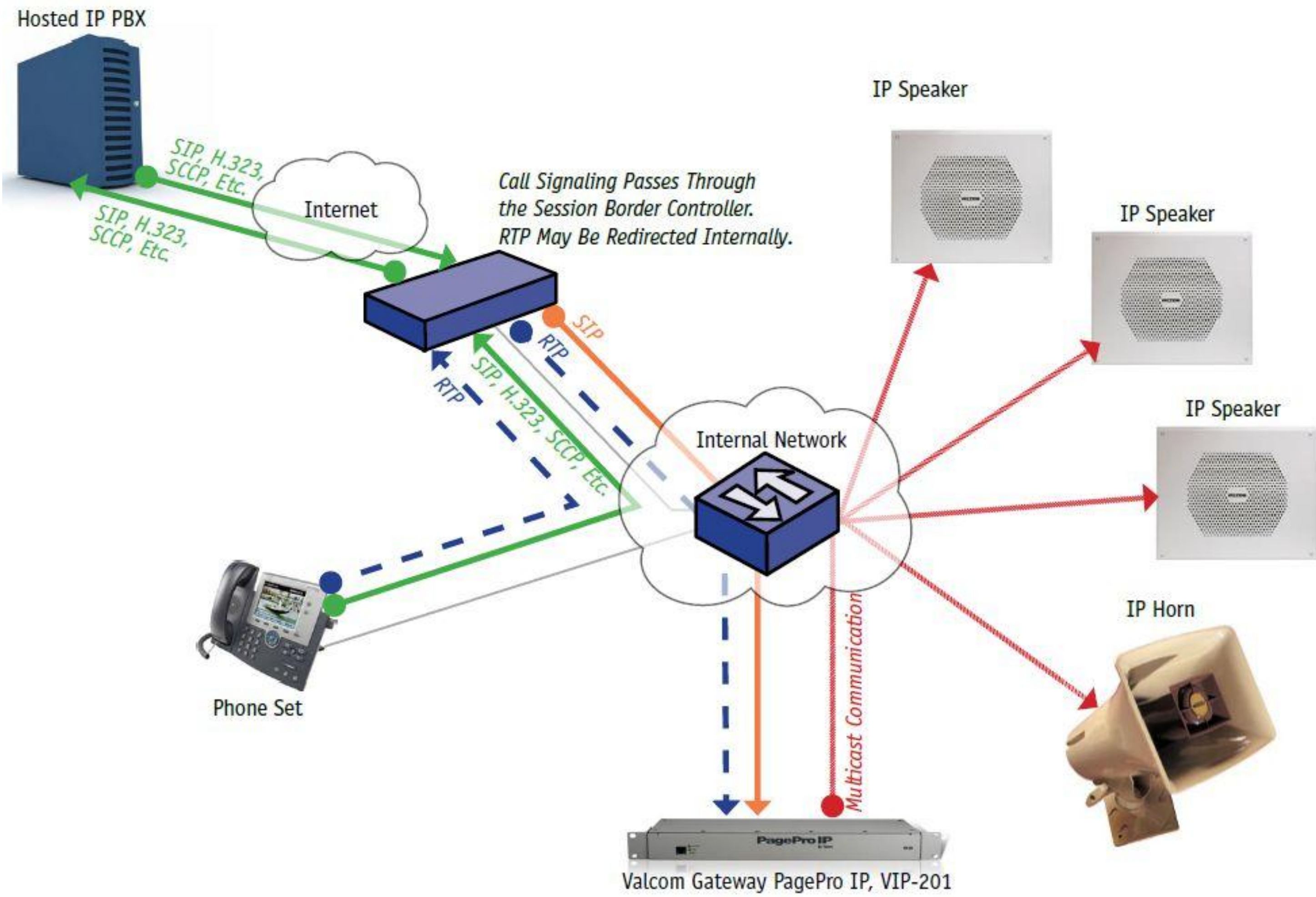


Direct SIP calls occur when the two endpoints send SIP signals directly to each other to initiate a call, without involving a SIP server. Valcom SIP devices support this method of calling. This is more likely to be seen with telephones that can assign line appearances to multiple SIP registrars.



SIP Trunk calls are similar to B2BUA calls, in that the SIP server is usually in the data stream. Route patterns must be created to send the calls to the correct trunk. Valcom SIP devices support this method, and this can be useful for older SIP servers that only supported SIP on their trunks, not as phones (Cisco CallManager 4, etc). Each Valcom endpoint must be defined as an individual trunk, but multiple SIP phone numbers can be handled on the trunk. A multi-channel device, such as a VIP-204, would only need one trunk definition.

SESSION BORDER CONTROLLER CALLS
(SBC)



Calls to hosted telephone systems will most likely use a Session Border Controller to provide the boundary between the customer's internal network and the service provider's network (or the Internet). A primary responsibility of the SBC is to modify the SIP information as it passes through the SBC to allow the call to connect through the firewall.