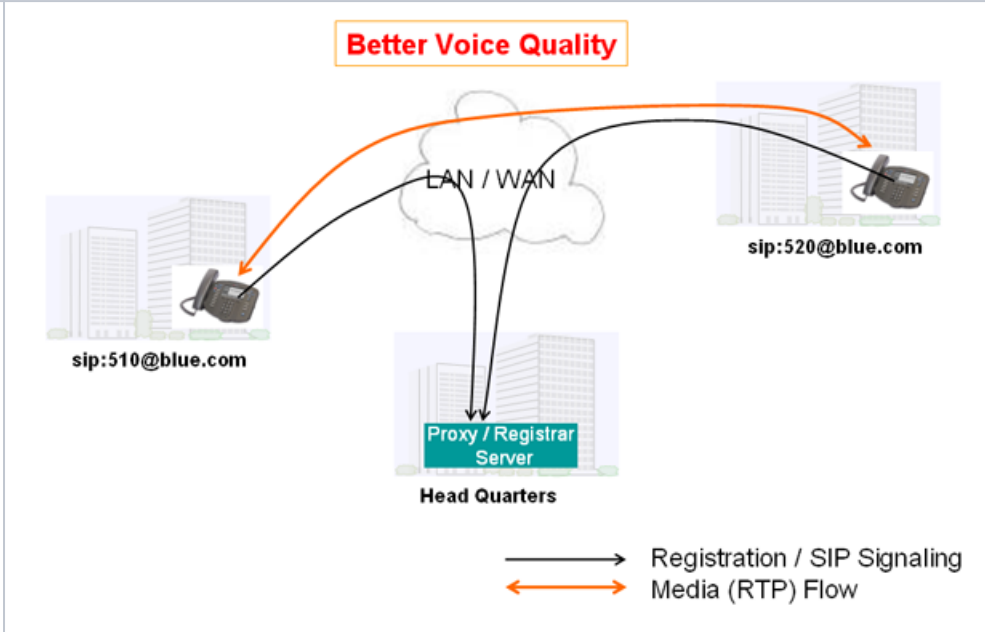


# Better Voice Quality

## sipXcom Offers Superior Voice Quality

The theory is easy: You move from a TDM solution to VoIP and expect the same voice quality. This article attempts to highlight some of the difficulties experienced in real deployments. You might also wonder why so many other IP PBX systems claim to be SIP standards compliant but violate the most basic principle of SIP: **The separation of signaling and media**.

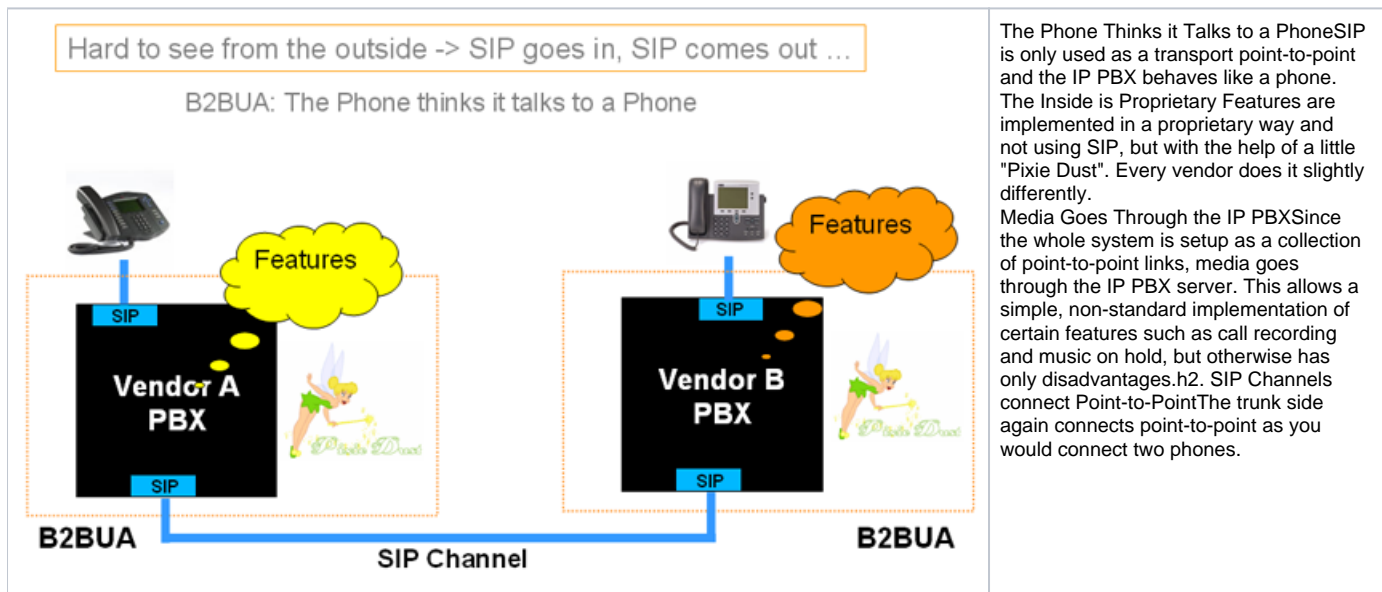
Consider the following example You have a headquarter location and two branch offices. The IP PBX is located at headquarters. A call is made from an extension at branch office 1 to an extension in branch office 2. The call setup signaling is handled by the IP PBX at headquarters, but where does audio go? "If your audio follows the same path as signaling does, you are in trouble. If audio has to go in and out of the IP PBX server, your trouble just got worse."  
Media is Peer-to-Peer Using SIP media was always meant to follow the most direct path possible. This is the only way you experience the least amount of delay, jitter, or packet loss. Other benefits include: \* Unlimited number of calls\* Codec negotiation between end points\* Support for all media\* HA failover works without disconnecting the call How comes so few IP PBXs do it this way?



While the separation of signaling and media seems an obvious requirement in the example above, it is also critical if a VoIP deployment is on just one local LAN. By definition, **call quality can only get worse** if the media stream is funneled through additional systems. The network interface used gets congested and therefore **limits the total number of simultaneous calls** that can take place to about 50 to 75. CPU cycles in your IP PBX server get tied up unnecessarily. A **single point of failure** is created where all the calls are disconnected should the PBX server fail. And finally, the IP PBX server has to **mediate during the codec negotiation phase** during call setup.

The reason for all this unnecessary complication and trouble can be found in the way **first generation IP PBX systems** are typically built.

## First Generation IP PBX Architecture



The Phone Thinks it Talks to a Phone SIP is only used as a transport point-to-point and the IP PBX behaves like a phone. The Inside is Proprietary Features are implemented in a proprietary way and not using SIP, but with the help of a little "Pixie Dust". Every vendor does it slightly differently. Media Goes Through the IP PBX Since the whole system is setup as a collection of point-to-point links, media goes through the IP PBX server. This allows a simple, non-standard implementation of certain features such as call recording and music on hold, but otherwise has only disadvantages. SIP Channels connect Point-to-Point The trunk side again connects point-to-point as you would connect two phones.

By far not the only example is Asterisk but, because of its open source model, it is easy to examine. Based on a Back-to-Back User Agent (B2BUA) design, all links are point-to-point, making phones believe they talk to another phone even though they talk to the PBX. The PBX becomes a big monolithic block where everything has to go through and features are implemented in a proprietary way. While in the TDM world these PBXs were built on expensive fault tolerant HW, using an inexpensive PC server does not serve this model well as it lacks scalability, resiliency and fault-tolerance. IAX, the Asterisk trunking protocol is specifically designed to bundle signaling and media in one fixed connection. The result is at best neutral when it comes to voice quality, but more often than not it is worse than what you would get with standard SIP.

## Second Generation IP PBX Architecture

IMS for the EnterpriseA SIP based application router (SIP Proxy) is the heart of the system. Strict separation of signaling and media is a given. The phone talks to a proxy that routes the call through as many additional proxies as needed until the destination phone is reached.

The Phones Negotiate CodecThe phones negotiate a selection of codec directly and the media stream is routed along the most efficient route - better voice quality and a more resilient connection are direct consequences.

Features are implemented in Feature ServersInside the IP PBX every feature is implemented as a SIP feature server. These feature servers can be co-located in the same physical host or distributed across many systems.

High-Availability by DesignFailover redundancy and system resiliency is simple to realize. Similar to the way web server farms load balance, so do SIP proxy servers.

