

YOUR MIGRATION TO VoIP TELEPHONY IS ALREADY NEARLY COMPLETE.



SIP HANDSET GATEWAY OVERVIEW

The components for a successful, cost-effective migration to VoIP are already up and running in your enterprise: Your existing wiring. Your existing LAN infrastructure. And, most importantly, your existing digital PBX and analogue handsets.

Many VoIP deployments require enterprises to “rip and replace” existing infrastructure, perform costly LAN assessments and upgrades, and overhaul cabling between the network room and desktop handsets. These aspects of a forklift VoIP upgrade can add thousands to total deployment costs and cause disruption to business operations. All before users benefit from the next-generation functionality and productivity enhancements of IP telephony.

Using the SIP Handset Gateways to “De-Obsolete” one of the most crucial components of your voice platform - desktop handsets. Your PBX handsets are already installed and employees are trained to use them. Your mission-critical database and web applications are up and running. The SIP Handset Gateways enable existing digital and analogue handsets with feature-rich, next-generation converged VoIP features and applications via a hosted or premise IP PBX. And it happens without costly network upgrades, business disruption, or additional training.

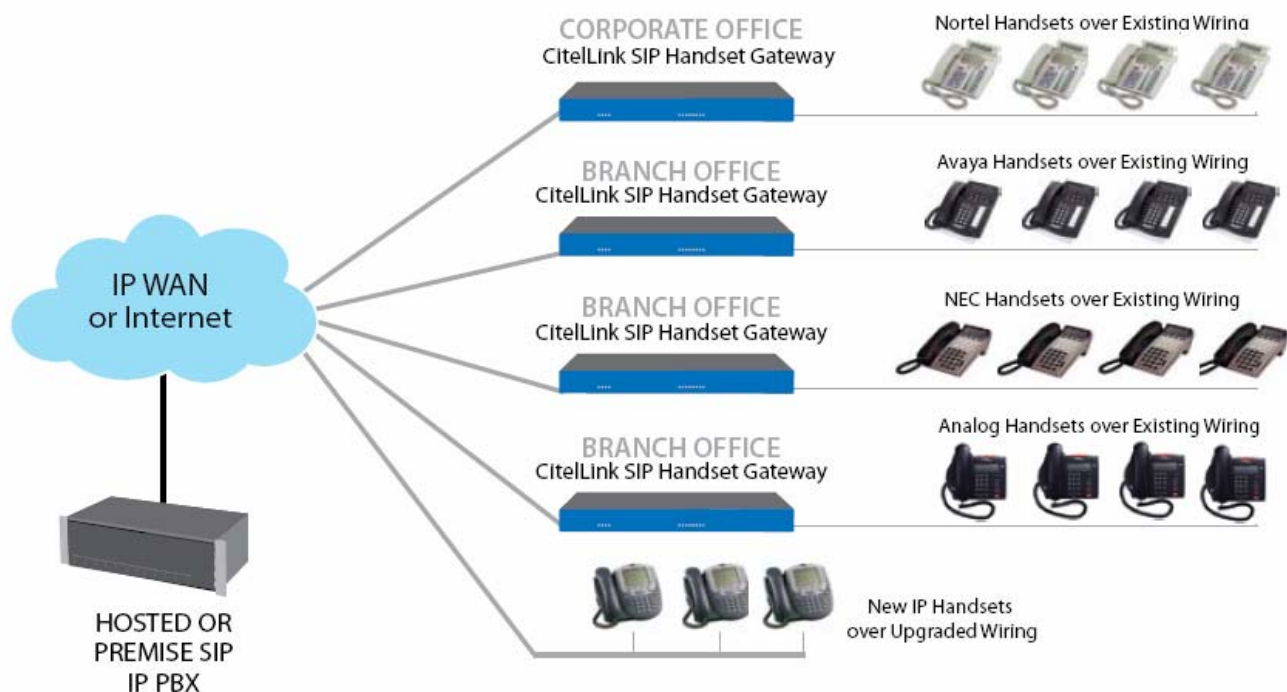


Deploy your VoIP platform quickly and with minimal business disruption. The SIP Handset Gateways are designed to make the migration to a hosted or premise IP PBX fast and worry free. With simple installation and programming through our intuitive GUI, many enterprises cut over to their VoIP platform in as little as five minutes.

Now, the business case for VoIP migration is clear.

No enterprise should deploy VoIP just for VoIP's sake. But the cost, convergence, and increased productivity benefits of IP telephony are clear. With a new hosted or premise IP PBX, The SIP Handset Gateways enable next generation call control, reporting and application convergence, with significant cost savings over other VoIP solutions. All while utilizing much of the infrastructure already deployed in your enterprise.

Sample Network Configuration



Overview Specifications

<p>Description SIP-Enabled Digital and Analog Terminal Adapter</p> <p>Capacity Digital: 12- or 24-Port Analog: 24-Port</p> <p>Supported Handset Models Over 60 models, including: Nortel Norstar Nortel Meridian 1 Nortel P Phone‡ Avaya Definity 64xx/84xx NEC Dterm i Ericsson** Panasonic* Toshiba* Analog (FXS)**‡</p> <p>IP Server Platforms Sylantro r.3.1 & later Broadsoft r.10 & later</p>	<p>IP-PBX Interoperability Asterisk Pingtel Zultys Avaya CS 3.0 & later</p> <p>Network Interfaces 10/100 BaseT Ethernet RS-232 FXO Analog*</p> <p>Supported VoIP Protocols SIP (per IETF-SIP-RFC3261) RTP</p> <p>VoIP Enhancements BLA (Bridge line appearance)</p> <p>Voice Support G.711 G.726* G.729*</p>	<p>Configuration & Management SNMP DHCP Web GUI Telnet*</p> <p>Serial port</p> <p>Physical <u>12-Port</u> 1U Rack Mountable 17" (432mm) W x 1.75"(44mm) H x 8"(203mm) D Weight: 6.4 lbs. (3kg) <u>24-Port, Digital & Analog</u> 1U Rack Mountable 19" (483mm) W x 1.75" (44mm) H x 17" (432mm) D Weight: 8.8 lbs. (4kg)</p>	<p>Power 90VAC-264VAC 50-60Hz 2.0A</p> <p>Environmental <u>12-Port</u> Operating Temp: 32° to 130° F (0° to +55° C) Humidity: 5% to 95% non-condensing <u>24-Port, Digital & Analog</u> Operating Temp: 32° to 104° F (0° to +40° C) Humidity: 5% to 85% non-Condensing</p> <p>Emissions FCC (Part 15, 68), CE , UL, EU, IEC, EN Certified</p> <p>*Digital 12-Port Model **Digital 24-Port Model ‡Analog 24-Port Model</p>
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