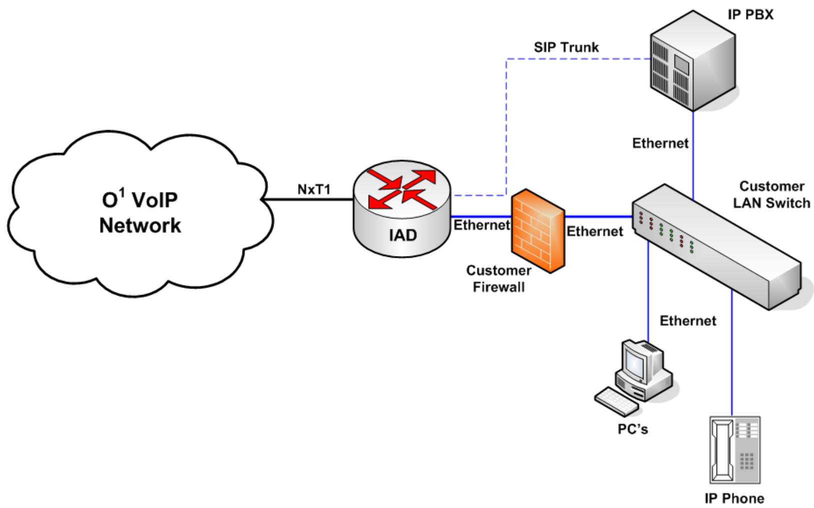
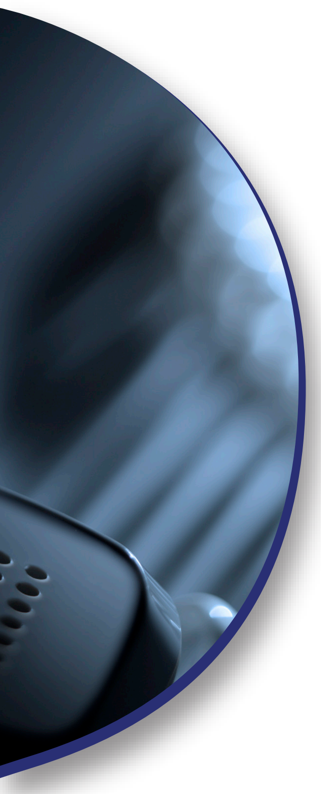
O1 VoiceStream SIP

Smart, Powerful and Cost-Effective

O1's Vo1ceStream Session Initiation Protocol (SIP) Trunking is the latest technology that is revolutionizing the way businesses make and receive telephone calls. This product works by linking the customers SIP-based premise equipment to O1's provided Integrated Access Device (IAD) through an IP connection. SIP eliminates the need for Public Switched Telephone Network (PSTN) media gateway and narrowband voice circuits.





Features

- From 5 up to 23 lines*
- Direct Internet Access up to 4.5Mbps*
- Up to 100 DIDs
- O1 managed Integrated Access Device
- Voice re-route to main backup telephone number
- 5 toll free numbers
- 4 foreign exchange DIDs
- Up to 8 IP addresses (5 usable)
- Web hosting
  - 50M storage
  - 50 GB transfer
  - 1 Domain name
  - 50 Email addresses
- 1 Basic white page listing
- Supports G.711 and G.729 codec
- Free inter-office calling among multiple locations

** Requires bonded circuits*

Benefits

- Unlimited local and IntraLATA calls
- Dynamic bandwidth management eliminates the need to maintain separate lines for voice and data
- Prioritized voice traffic for voice quality, QoS
- No charge to port over existing numbers
- No need to buy or maintain local PSTN gateway
- The SIP solution grows with your business
- Long distance rate plans starting at a minimum of 1000 minutes per bundle
- 24 x 7 dedicated support
- MPLS optional add-on

Customer Phone System Requirements

- SIP RFC 3261 standard
- Ability to revert to G.711 if G.729 route fails
- Accept periodic "heartbeat" of Re-invite
- Accept OPTIONS pings every 60 seconds

O1 Communications, Inc. 1020 16th Street, 2nd Floor, Sacramento, CA 95814 T. 888.444.1111 F. 916.554.2180
www.o1.com