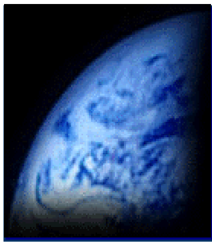


Centro VOIP PBX Telephone System
Starter Package

What does the price include?

- Dell™ Hi Performance Server PC with 1 year warranty
- Trixbox® CE 2.6.2 pre-installed (Elastix®, AsteriskNOW®, PIAF® also available)
- Pre-configured OpenVox A400P01 card (4port FXO module) with 5 year warranty
- Grandstream GXP2000 Voip telephone handset
- 2GB USB 2.0 external flash drive
- 2 Meter RJ-11 <-> BT crossed modem cable (UK customers only)
- Server handbook / guide
- 30 day email / telephone installation support

Advantages of a Pre-Built Trixbox® Certified Server

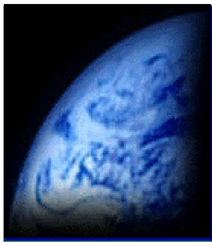
The Dell Hi Performance server PC comes preloaded and fully tested with Trixbox® CE 2.8.0. and OpenVox A400P card have been fully tested/verified for compatibility with Trixbox® by Fonality, the company behind trixbox®. The main advantages of purchasing a pre-built Trixbox® certified server are:

- Reduces testing/configuration time and lowers deployment risks
- Enables resellers to focus on deploying customer solutions and developing their business
- Allows end customers installing their own phone system to start setting up the system straight away (i.e. adding extensions, configuring the dial plan / IVR, etc.) instead of wasting time troubleshooting software/hardware installation issues
- The pre-configured OpenVox A400P01 card (1 FXO port) means that the box is ready to go out of the box
- Automated configuration file backup to 2Gb USB flash drive provides extra protection

Product Overview

The Trixbox® Certified Dell is a reliable, highly scalable server designed to address the needs of small to medium-sized businesses. The delivers the latest performance features at an aggressive price point to provide exceptional value for money. With a fast Intel Celeron processor, a Gigabit of RAM and dual 80GB Raid 1 storage, the Dell trixbox® CE 2.8.0 based IP PBX server provides a fully featured PBX with support for up to 15 simultaneous calls and 60 users/extensions.

The server comes with a pre-configured Trixbox® Certified OpenVox A400P01 card, which provides Four FXO port for connecting to 4 PSTN lines. Using a SIP/IAX2 VoIP trunk in addition to a PSTN line



Centro VOIP PBX System

Provides the optimum solution in terms of low cost call routing and reliability. Using a VoIP trunk as the primary route for outbound calls minimises call costs, and having a PSTN line connected provides a reliable backup in the event of an Internet connectivity issue.

Asterisk® based telephony solutions offer a rich and flexible feature set. They provide both classical PBX functionality and advanced features, which interoperate with traditional telephony systems and Voice over IP solutions. Trixbox® is the world's most popular distribution of Asterisk® with over 65,000 downloads per month. Trixbox® CE provides extensive flexibility to satisfy the needs of standard and bespoke customer deployments. As an alternative to Trixbox®, the server can also be preloaded with Elastix®, AsteriskNOW®, or PIAF® if preferred.

Hardware Features

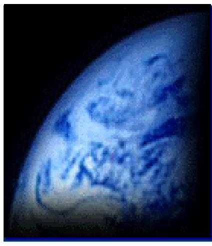
- 1Gb of fast 667Mhz DDR II memory to increase performance and avoid use of hard disk swap space
- Server class Error Correcting Code (ECC) memory to detect and correct data errors caused by the storage/transmission of data
- Raid 1 disk mirroring to protect against hard disk failure
- Pre-configured automated configuration file and voicemail backup to an external USB flash drive (included) to provide extra protection
- Large 512K L2 cache to minimise memory access
- Fast 800Mhz Front Side Bus (FSB) to minimise data transfer times
- Highly scalable with 2 x PCI slots, 3 x PCI-E slots (x8, x4, x1)
- Processor support for NX (No eXecute) bit to increase system security by allowing areas of memory to be marked as non-executable
- Accurate timing - zttest results after 60 passes, Best: 100.000, Worst: 99.994, Average: 99.995755

System Customisation Options

The telephone system advertised includes the IP PBX server itself and an OpenVox A400P01 card for connecting the IP PBX server to a PSTN line. The server can be upgraded and phone handsets can be added to your order at a discounted price to provide a full telephony system. Some of the options available include:

- Server upgrades, e.g. faster CPU, more memory, etc.
- Alternative Asterisk Package, e.g. Elastix®, AsteriskNOW®, PIAF®
- Additional analogue FXO/FXS ports
- ISDN BRI and PRI ports
- VoIP SIP/IAX2 phone handsets
- Professionally recorded IVR prompts and music on hold scripts
- VoIP SIP/IAX2 FXS ATA/Gateways, e.g. for remote/home workers

To request a quote for a customised system please email us at sales@centro.co.uk.



Installation Requirements

- Broadband Internet Connection for VoIP trunks, software package updates, remote access, etc.
- Spare router or switch port to provide LAN / Internet connectivity
- Static IP Address is recommended although not essential, e.g. Dynamic DNS can be used
- Laptop or PC to use for configuring the Trixbox® server and any VoIP phone handsets or gateways
- Basic networking, Internet telephony and Linux knowledge/skills

The Trixbox® server comes pre configured with default settings and passwords except for country specific voice prompts (where available) and PSTN line settings for the FXO port.

Server Handbook / Guide

A server handbook / guide will be supplied to provide all the information required to ensure the server can be fully managed/supported. The Trixbox® IP PBX handbook will include the following information:

- Interface and network IP address settings
- Username/password information
- Hardware information including model/serial numbers
- Software information including version numbers
- PCI card configuration settings
- Show command information, e.g. PCI interrupts, card information, etc.
- Troubleshooting tips/guidelines
- Centro support contact details and warranty information

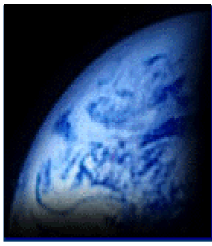
Support Options

The Trixbox® IP PBX comes with 30 days email/telephone support for installation issues included. The pre-configured FXO port to connect to the PSTN is guaranteed to work out of the box. In the event of a PSTN connectivity issue full support will be provided to solve the issue (remote access may be required).

It is important to understand that setting up an IP PBX can be a complex task depending on the deployment requirements, e.g. dial plans, IVR menus, email to voicemail, local/remote extensions, SIP/IAX2 trunks, etc. Multiple paid for support options are available including:

- Pre configuration before despatch (e.g. dial plans, IVR menus, extensions,etc.)
- Full on site system installation (half a day or full day on site depending on customer requirements)
- remote and on site support contracts
- Remote offsite backup for configuration files and voicemail

To request a quote for support please email us at sales@centro.co.uk.



Centro VOIP PBX System

Trixbox® IP PBX Software Features

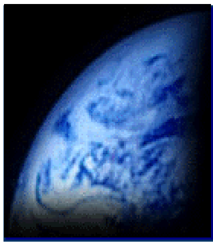
Trixbox® CE 2.8.0 provides an extensive feature set, some of the features supported are listed below:

- Analogue FXO Trunks (1 pre-configured FXO included)
- ISDN BRI / PRI Trunks
- VoIP SIP / IAX2 Trunks
- Analogue FXS and VoIP SIP / IAX2 Extensions
- Automated Attendant
- Blacklists
- Call Forwarding, Monitoring, Recording
- Caller ID and Caller ID Blocking
- Direct Inward System Access
- Free Calls to Remote Extensions
- Voicemail
- Fax Support
- Voicemail to Email
- IVR Menu System
- Ring Groups
- Call Queues
- Conference Rooms
- Follow-Me
- Time-Based Routing
- Music On Hold
- Paging and Intercom
- Web Access to Voicemail
- Admin Status Screen
- Package Manager (for easy updates)
- Phone Provisioning Tool (Endpoint manager)
- Network Settings Tool
- Enhanced CDR Reports
- Echo Cancellation - OSLEC (Open Source Line Echo Cancellation)

Server Specification

The server specification can be upgraded if required, please email us at sales@centro.co.uk with your requirements before placing your order. The standard specification is as follows:

- Intel® Celeron® processor 430, 1.8GHz, 512KB L2 cache, 800MHz FSB
- 1GB memory, DDR2, ECC, 667MHz
- Dual 80GB (RAID 1) SATA II hard disk drives, 7200rpm, 8MB Cache
- 48X CD / 18X DVD-ROM IDE/PATA drive
- Single embedded Gigabit Ethernet NIC
- 2GB USB 2.0 external flash drive, 10MB/s read, 8MB/s write
- OpenVox A400P 4-port analogue PCI card with one FXO module
- Server preloaded with trixbox® CE 2.6.2, which is based on CentOS 5.2, Zaptel 1.4.12.1, and Asterisk 1.4.22



Centro VOIP PBX System

OpenVox A400P11 + Grandstream GXP2000

+

The Grandstream GXP2000 next-generation SIP

Phone features include 4 SIP lines, 7 speed dial keys, multiline support for up to 11 lines, 8 line 22 character LCD display with blue backlight, and dual switched 10/100Mbps autosensing ports with PoE.

TOTAL COST £595.00

Optional Installation for up to 6 extensions **£400.00**

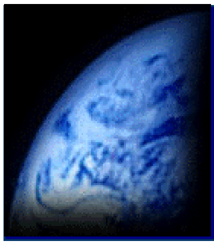
***Line and Broadband Options**

Analogue line installation £99.00 Monthly rental £12.50
Basic rate ISDN line installation £300.00 (2 channels) Monthly rental £28.00
Broadband max circuit with 8mb download 448k Upload with static IP address £24.99 per month.

Draytek Business Class Broadband Router Cost £195.00

*** line installation and monthly charges may vary**

All Prices Quoted are exclusive of VAT and DELIVERY



Centro VOIP PBX System

ADDITIONAL HANDSETS

Grandstream Budgetone 201

The Grandstream Budgetone 201 offers a family of affordable, next generation SIP phones that features excellent audio quality and rich telephony features. The BudgeTone Series supports popular voice codecs and is designed to be fully interoperable with 3rd party SIP providers.

The BudgeTone phones have upgraded features including 3-way conferencing, full-duplex hands-free speakerphone (BT-201), voicemail with indicator and custom ring tones. The BudgeTone SIP Phone Series is an affordable, easy to use VoIP solution for the home or SoHo office.

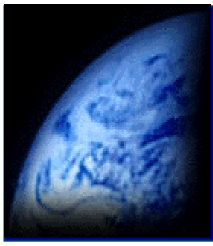
Grandstream BudgeTone 201 IP Phone with 3-Way Conferencing

The Grandstream 201 is the most powerful phone in the BudgeTone series. It offers upgraded features including 3-way conferencing, full-duplex hands-free speakerphone, voicemail with indicator and custom ring tones.

The BudgeTone 201 is an affordable, easy to use VoIP solution for the home or office.

Grandstream BudgeTone 201 Features

- Interoperable & compatible with SIP platforms
- Single (1) 100 Mbps auto sensing Ethernet RJ45 ports
- Full duplex speakerphone
- Standard voice features and functionality
- Voicemail indicator with light



Centro VOIP PBX System

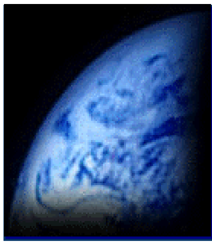
The Grandstream GXP-2000

The Grandstream GXP-2000 is a modern business IP phone that utilises open technology standards. Created with leading edge technologies, the GXP2000 provides innovative and excellent sound quality, extensive features and great configurability at amazing prices.

The Grandstream GXP2000 is expandable, secure and easy to manage, offering 4 individual SIP accounts, 7 programmable keys, visual message indicator, full duplex hands-free speakerphone, dual 10M/100Mbps Ethernet ports, intuitive user interfaces, large back-lit graphical LCD display, security and privacy protection, XML screen content customisation, automated phone book synchronisation with directory server using XML, as well as broad interoperability with most 3rd party SIP products.

The GXP2000 is an ideal IP phone for both the small business and the enterprise customer.

- Advanced jitter buffer control, packet delay and loss concealment technology
- Support for popular codecs including G.711, G.726, G.728, G.723, G.729, GSM, G.722 (wideband) and ILBC with dynamic negotiation of codec type and packet time
- Supports 4 independent SIP accounts or SIP Server platforms with 11 line indicators (expandable to a 123 more through expansion key-module)
- Supports 7 programmable keys
- Large 131x64 Graphical LCD to display up to 8 lines and 22 characters per line
- Full duplex speakerphone with advanced echo cancellation
- Dual 10/100Mbps Ethernet ports
- Headset jack
- Supports Caller ID display or block, per call or permanent
- Supports Call waiting, Hold, Mute, Transfer (blind or attended), Forward (on busy, unconditional or no answer) , and more
- Supports Multi-party conferencing, Intercom and various DTMF options (RFC2833, SIP INFO, in audio)
- Supports Integrated Power-over-Ethernet (802.3af and Cisco style)
- Message Waiting Indicator for voicemail
- Call Log
- Phone Book
- **COST £75.00**



Centro VOIP PBX System

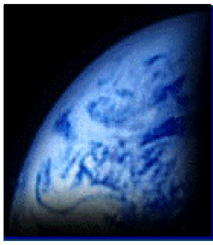
The Grandstream GXP-1200

Product Overview

The Grandstream GXP1200 next-generation entry-level SIP VoIP Phone offers the same market-leading superb sound quality and rich functionality as the award winning GXP2000 . The GXP1200 features include 2 line appearance with dual coloured LED indicator and 2 independent SIP accounts, 128x32 pixel screen with blue backlight, XML programmable context sensitive soft keys, dual switched 10/100 autosensing network ports with integrated PoE, and a full duplex speakerphone with advanced acoustic echo cancellation.

Feature Summary

- 2 SIP line indicator keys and mute button, 3 XML programmable context sensitive soft keys, 5 navigation/menu/volume keys
- 8 Dedicated Feature Keys: Message, Hold, Transfer, Conference, Headset, Speakerphone, Redial/Send, Mute/Del
- Voice Features: Downloadable Phone Book (XML, LDAP up to 200 items), XML Customisation of Screen, Voice Mail Indicator, Redial, Call Log, Volume Control, Caller ID Display or Block, Call Waiting, Hold, Transfer, Forward, FLASH, Mute, 3-Way Conferencing, off-hook auto dial, configurable emergency dialling (e.g., 911), early dial, click-to-dial, auto answer, downloadable ring tones
- Dual RJ-45 10/100 autosensing ports (switched or routed) with integrated PoE to allow a desk PC to be connected to the phone in order to conserve switch/router ports and AC sockets in order to simplify deployment
- NAT router and DHCP server
- Multi language support (English, German, Italian, French, Spanish, etc.)
- Secure Real-time Transport Protocol (SRTP) and SIP Transport Layer Security (TLS) (pending) support
- Full duplex handsfree speakerphone with Advanced Echo Cancellation (AEC), Acoustic Gain Control (AGC) and side tone support
- Easy configuration through manual operation (phone keypad and Web interface) or personalized automated provisioning via central configuration file for mass deployment.
- NAT-friendly remote firmware upgrade capability via tftp/http even from behind firewalls/NATs
- Interoperable with various 3rd party SIP end user device, Proxy/Registrar/Server, and gateway products (e.g., Asterisk®/Trixbox® IP PBX, MS Messenger, Cisco® IP phone and gateway, etc)
- Dynamic negotiation of codec and voice payload length
- SIP and DNS server redundancy and failover
- Features Advanced Digital Signal Processing (DSP) to ensure superb hi-fidelity audio quality
- Advanced and patent pending adaptive jitter buffer control, packet delay and loss concealment technology



Centro VOIP PBX System

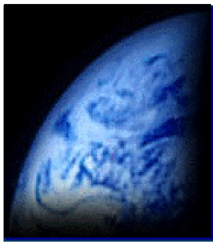
Technical Summary

- IEEE 802.3af standards based Power Over Ethernet (PoE) support
- Voice codecs supported include G.723.1 (5.3K/6.3K), G.729A/B, G.711 (a-law and u-law), G.722 (wide-band), G.726 (32K), GSM and iLEB codecs.
- Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- In-band and out-of-band DTMF (in audio, RFC2833, SIP INFO)
- Support for Fixed IP, DHCP, SIP Presence (SIMPLE), Automated NAT traversal using IETF STUN (manual configuration of firewall/NAT not required) and symmetric RTP (compatible with Cisco's ATA-186, etc)
- Protocols supported include SIP RFC3261, TCP/UDP/IP, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS (A record and SRV), DHCP (client and server), NTP, TFTP, PPPoE
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- DIGEST authentication and encryption using MD5 and MD5-sess

Hardware Summary

- LAN interface: 2 x RJ45 10/100Mbps autosensing
- Headset Jack: RJ-11 Headset port
- LED: 1 dual coloured LED, 2 line indicators and mute button
- Phone Case: ABS plastic, 30-button keypad, comes with detachable footstand and spacers for wall mounting
- Display: 128x32 pixel graphic LCD with blue backlight
- Universal Switching Power Adaptor: Input: 100-240VAC; 50-60 Hz, Output: +5VDC, 1200mA, UL certified
- Dimensions (excluding stand): 195mm (W) x 201.7mm (L) x 77.5mm (H)
- Handset Weight (excluding stand): 730g
- Package Weight: 1325g
- Temperature: 32 - 104 ºC (0 - 40 ºF)
- Humidity 10-90% (non condensing)
- Compliance: FCC / CE / C-Tick / RoHS

Cost £65.00



Centro VOIP PBX System

The Grandstream GXP-2000 Extension Module

Product Summary

The Grandstream GXP2000 Extension Module provides 56 fully programmable speed dial buttons, each with a dual colour LED. The GXP2000-EXT is controlled by the GXP2000 IP Phone which it connects to via a short connector cable. The GXP2000-EXT creates up to 112 additional programmable buttons when 2 EXT modules are daisy chained with the GXP2000. The GXP2000-EXT supports speed dialling, Busy Lamp Field (BLF), Bridged Line Appearance (BLA), call transfer/forward/pickup on each of the programmable buttons.

Hardware Summary

- LED: 56 dual colour line indicator LEDs and 1 message indicator LED
- Extension Module Case: ABS plastic, 56-button keypad
- 6-pin Mini-DIN (PS/2 type) connector cable and steel connector back plate
- Universal Switching Power Adaptor: Input: 100-240VAC; 50-60 Hz, Output: +5VDC, 1200mA, UL certified
- Dimensions: 193mm (W) x 215mm (L) x 57mm (H)
- Extension Module Weight: 770g
- Package Weight: 1130g
- Temperature: 32 - 104 ºC (0 - 40 ºF)
- Humidity 10-90% (non condensing)
- Compliance: FCC / CE / C-Tick / RoHS

<u>Cost £48.99</u>