

# Open Source PBX

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## Background

- Budget cuts
- Assortment of phone equipment and systems
- Lack of Scalability

## Design Goals

- Cut phone operating budget
- Better service for less money
- Scalability
- Leverage existing infrastructure

## Why Asterisk?

- Open source
- Integration with POTS
- Widely used and supported
- Runs on commodity hardware
- Open Standards - SIP

## How we got there

- July 2006 - LCSR prototype
- Sept. 2006 - DCS turns off phones for faculty and students
- Oct. 2006 - Voicemail-to-email system
- May 2007 - Deployment begins
- Aug. 2007 - Deployment complete

## Features

- Voicemail-to-email
- Phone directory
- Ring patterns
- Call restrictions

## Problems

- POTS problems
- Echo
- 911
- Learning curve - <http://freepbx.org>

## Where do we go from here?

- Connecting VoIP “islands” - connecting departments
- Long Distance
- SIP.edu -  
<http://www.internet2.edu/sip.edu/>
- Soft phones

## More Info

- Asterisk - <http://asterisk.org>
- Digium - <http://digium.com>
- FreePBX - <http://freepbx.org>
- SIP.edu - <http://www.internet2.edu/sip.edu/>
- VoIP Wiki - <http://www.voip-info.org/wiki/>
- Snom - <http://snom.com>
- DCS VoIP project -  
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