Why?

The number one reason to switch to VoIP technology for telephone service is cost reduction. From that base, VoIP is able to provide some compelling features which makes switching even more attractive.

Eliminating Phone Lines

With VoIP service, you can cancel your traditional phone service through your local telephone company and place all of your telephone calls over your broadband Interner connection.

For a residential customer, this will save around $40 a month. For business customers, the savings can be thousands of dollars a month.

Eliminating Long Distance Charges

VoIP technology can also save money on long-distance charges. Most residential and business telephone customers pay per-minute fees for long-disatance telephone calls. VoIP can reduce or eliminate those long-distance fees.

This saving is especially valuable with International calls, where per-minute charges for traditional telephone calls can be very expensive.

Computer Telephony Integration (CTI)

VoIP service providers are designing and implementing new features which implement Computer Telephony Integration (CTI).

For example, VoIP customers may be able to receive their voice messages in e-mail as .WAV file attachments. This can make managing voice mail messages much easier and more powerful, because it enables recipients to archive voicemails or forward them to anyone with an email address.

Why Woodbridge Data?

Simply put, we practice what we preach. We embraced this technology since our inception and utilize our own self-branded VoIP PBX system in-house. It is identical in concept and execution to our offering to our customers.

We have the design and installation experience and the ability to leverage our relationships with many data services providers (Comcast, Cablevision, Time Warner, FIOS, and any DSL provider you select) to make your VoIP PBX solution easy to use, quick to install, and absolutely reliable.

Our systems have the following features:

Call Features

ADSI On-Screen Menu System
Alarm Receiver
Append Message
Authentication
Automated Attendant
Blacklists
Blind Transfer
Call Detail Records
Call Forward on Busy
Call Forward on No Answer
Call Forward Variable
Call Monitoring
Call Parking
Call Queuing
Call Recording
Call Retrieval
Call Routing (DID & ANI)
Call Snooping
Call Transfer
Call Waiting
Caller ID
Caller ID Blocking
Caller ID on Call Waiting
Calling Cards
Conference Bridging
Database Store / Retrieve
Database Integration
Dial by Name
Direct Inward System Access
Distinctive Ring
Distributed Universal Number Discovery (DUNDi™)

Do Not Disturb
E911
ENUM
Fax Transmit and Receive (3rd Party OSS Package)
Flexible Extension Logic
Interactive Directory Listing
Interactive Voice Response (IVR)
Local and Remote Call Agents
Macros
Music On Hold
Music On Transfer:
- Flexible Mp3-based System
- Random or Linear Play
- Volume Control

Call Features

Predictive Dialer
Privacy
Open Settlement Protocol (OSP)
Overhead Paging
Protocol Conversion
Remote Call Pickup
Remote Office Support
Roaming Extensions
Route by Caller ID
SMS Messaging
Spell / Say
Streaming Media Access
Supervised Transfer
Talk Detection
Text-to-Speech (via Festival)
Three-way Calling
Time and Date
Transcoding
Trunking
VoIP Gateways
Voicemail:
- Visual Indicator for Message Waiting
- Stutter Dialtone for Message Waiting
- Voicemail to email
- Voicemail Groups
- Web Voicemail Interface
Zapateller

Computer-Telephony Integration

AGI (Asterisk Gateway Interface)
Graphical Call Manager
Outbound Call Spooling
Predictive Dialer
TCP/IP Management Interface

Scalability

TDMoE (Time Division Multiplex over Ethernet)
Allows direct connection of Asterisk PBX
Zero latency
Uses commodity Ethernet hardware
Voice-over IP
Allows for integration of physically separate installations
Uses commonly deployed data connections
Allows a unified dialplan across multiple offices

Codecs

ADPCM
G.711 (A-Law & μ-Law)
G.722
G.723.1 (pass through)
G.726
G.729 (through purchase of a commercial license)
GSM
iLBC
Linear
LPC-10
Speex

VoIP Protocols

SIP (Session Initiation Protocol)
IAX™ (Inter-Asterisk Exchange)
H.323
MGCP (Media Gateway Control Protocol
SCCP (Cisco® Skinny®)

Traditional Telephony Protocols

E&M
E&M Wink
Feature Group D
FXS
FXO
GR-303
Loopstart
Groundstart
Kewlstart
MF and DTMF support
Robbed-bit Signaling (RBS) Types
MFC-R2 (Not supported however, a patch is available)

PRI Protocols

4ESS
BRI (ISDN4Linux)
DMS100
EuroISDN
Lucent 5E
National ISDN2
NFAS
Q.SIG

Call us today to schedule an appointment to discuss your telephony needs (732) 669-0601.