

## **Telephony, Telephone, Voice, and Voice over IP technologies.**

### **Key System:**

#### **Lines**

Telephone number and “channel” are one and the same  
Put someone on hold on “line 2” and yell for someone else to pick up “line 2”

#### **PBX:**

**Trunks:** Inbound or outbound phone channels

**DIDs:** Direct inward dialing – Phone numbers

When you buy dids, you buy blocks in most cases, and buy extra  
For example, one company I worked for bought two entire central office prefixes of  
10,000 numbers each: 520-872-XXXX 520-873-XXXX

#### **Trunks:**

**FXO** – Single analog line from the telephone company

One number / one line relationship

Echo issues due to single pair configuration

**ISDN-BRI** – Two digital lines from the telephone company on a single pair

Little to no echo issues

128Kbit bonded internet/data capability

2B+D (Bearer channel 64kbit = digital phone line), (Data channel 16kbit = signaling)

**T-1 RBS** – T1 line with Robbed bit signaling

24 64Kbit channels, with 8bits robbed per channel for signaling

24 56Kbit usable channels, signaling is similar to analog, with tones per channel

AMI and B8ZS signaling channels

**T-1 ISDN-PRI** – 23 digital lines from the telephone company on one or two pairs (two pairs for hand over to customer as most CPE uses two pairs for T1/PRI)

Little to no echo issues

Supports DIDs due to the ISDN “D” channel for signaling

23B+D (Bearer channel 64kbit = digital phone line), (Data channel 64kbit = signaling)

Cost: \$350-800/month

**SIP Trunk** - Session Initiation Protocol (signaling) combined with RTSP (Real Time Streaming Protocol)

Digital channel over Internet Protocol – Normally with UDP packets  
Variable latency – aka Jitter can be problematic  
High latency – Normally will still work, though the users may notice it

**G.711** – Uncompressed digital trunk 80Kbit/sec with IP overhead

**G.729a** – Highly compressed digital trunk 24Kbit/sec with IP overhead

**GSM** – Highly compressed digital trunk – 29Kbit/sec with IP overhead

**G.722** – Highly compressed, HD or high fidelity codec 16KHz samples instead of the standard 8 KHz samples – 80Kbit/sec with IP overhead

IP Ports 5060 (setup) RTSP picks almost anything over 10,000

**IAX2 Trunk** - Inter Asterisk Protocol

Uses a single port, 4569, and is much more NAT friendly

Digital channel over Internet Protocol – Normally with UDP packets  
Variable latency – aka Jitter can be problematic  
High latency – Normally will still work, though the users may notice it

**G.711** – Uncompressed digital trunk 64Kbit/sec plus combined overhead

**G.729a** – Highly compressed digital trunk 8Kbit/sec plus combined overhead

**GSM** – Highly compressed digital trunk – 13Kbit/sec plus combined overhead

**G.722** – Highly compressed, HD or high fidelity codec 16KHz samples instead of the standard 8 KHz samples – 64Kbit/sec plus combined overhead

**VoIP over Public Internet:**

**SIP/IAX2**

Variable latency that you do not have any control over  
Some ISPs block traffic (more common in Mexico and Latin America)  
Not encrypted by default

Best for tying asterisk systems together due to reduced overhead  
Not many telephone companies support IAX2

## **VoIP over private line:**

### **SIP/IAX2**

Provider controlled line, with latency and up-time guarantees

Not encrypted, but it's direct between you and your provider, limiting who can be in the middle

### **Channel Bank**

Takes in either RBS or PRI T1 (RBS is fine), and provides 24 analog ports

Depending on the model can be configured for all FXO all FXS, or a mix, and some can do data break out as well.

**FXO** = Central Office

**FXS** = Telephone station

[http://www.asteriskguru.com/tools/bandwidth\\_calculator.php](http://www.asteriskguru.com/tools/bandwidth_calculator.php)