Voip #8

Voice Over Internet Protocol
Server System-Raspian

What is VoIP?

A communication tool that uses an IP address to dial into to phone systems.
-Internal Communication within a network
-External Communication to Landlines
-Ingoing calls from landlines with phone number direction services
-Commercial Example: Skype

Implementation

-Pre-built & open source systems exist for Linux and RasPi

-Hopefully we can use these as a guideline and extract the data tools we need.

-H.323 Comm Protocol Data Transmission

-Other possibilities:

Video Conferences

Free outgoing calls

10/18

GitHub user: n02018222/
rep: /10-18 *voip*

https://github.com/n02018222/test

http://www.youtube.com/watch?v=GCBROgPCox0

http://www.raspberry-asterisk.org/downloads/

http://www.raspberry-asterisk.org/documentation/#nextsteps

raspbx-upgrade

```
<div class="content">
<h1 class="articlePageTitle">Advantages of Using VoIP</h1>
<p>Click "Play" to see how packet switching works.</p><div class="line">
	<div class="center interactive" style="clear:both;height:300px;width:400px;">
		<object type="application/x-shockwave-flash" data="http://static.howstuffworks.com/flash/ip-telephony-packet.swf" width="400" height="300" id="id_5260ad09ef0af" style="visibility: visible;"><param name="allowScriptAccess" value="always"><param name="wmode" value="opaque"><param name="bgcolor" value="#FFFFFF"></object>
	</div>
</div>
<script type="text/javascript">
/* <![CDATA[ */
(function()
{HSW.interactive({flash:'http://static.howstuffworks.com/flash/ip-telephony-packet.swf',flashVars:{},flashParams:{},id:'id_5260ad09ef0af',width:'400',height:'300',aditure:{rule:'timedEvent'}});})();
/* ]]> */
</script>

	</div>
```

http://static.howstuffworks.com/flash/ip-telephony-packet.swf

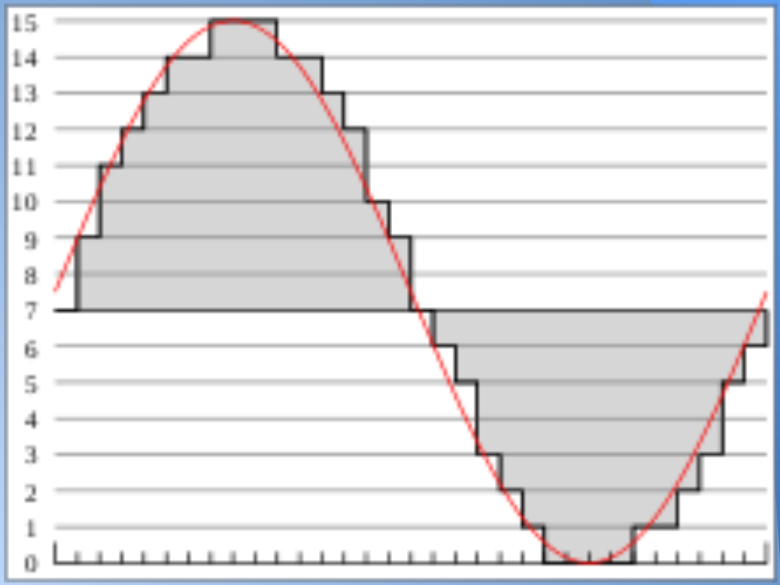
Here's how a typical telephone call works:

1. You pick up the receiver and listen for a dial tone. This lets you know that you have a connection to the local office of your telephone carrier.
2. You dial the number of the party you wish to talk to.
3. The call is routed through the switch at your local carrier to the party you are calling.
4. A connection is made between your telephone and the other party's line using several interconnected switches along the way.
5. The phone at the other end rings, and someone answers the call.
6. The connection opens the circuit.
7. You talk for a period of time and then hang up the receiver.
8. When you hang up, the circuit is closed, freeing your line and all the lines in between.

via: http://computer.howstuffworks.com/ip-telephony2.htm

Codecs accomplish the conversion by **sampling** the audio signal several thousand times per second. For instance, a G.711 codec samples the audio at 64,000 times a second. It converts each tiny sample into digitized data and compresses it for transmission.

When the 64,000 samples are reassembled, the pieces of audio missing between each sample are so small that to the human ear, it sounds like one continuous second of audio signal. A G.729A codec has a sampling rate of 8,000 times per second and is the most commonly used codec in VoIP.



The most widely used protocol is **H.323**, a standard created by the International Telecommunication Union (ITU). H.323 is a comprehensive and very complex protocol that was originally designed for **video conferencing**. It provides specifications for real-time, interactive videoconferencing, data sharing and audio applications such as VoIP.