HOW TO BUILD YOUR OWN PBX

By

Using Asterisk

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Overview

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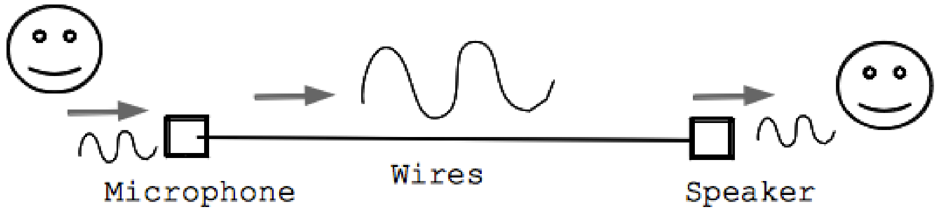
- SIP Call Setup
- SIP Responses
- DEMO

Introduction To VOIP

- ## Analog Phone System

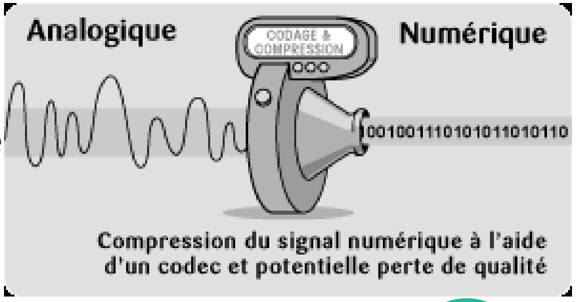


- Person talks into microphone
- Microphone converts sound waves into electrical waves
- Electrical waves go down the wire to the other phone

- At the other phone, speaker converts the incoming electrical wave back into sound waves Analog.

- ## Convert from Analog to Digital

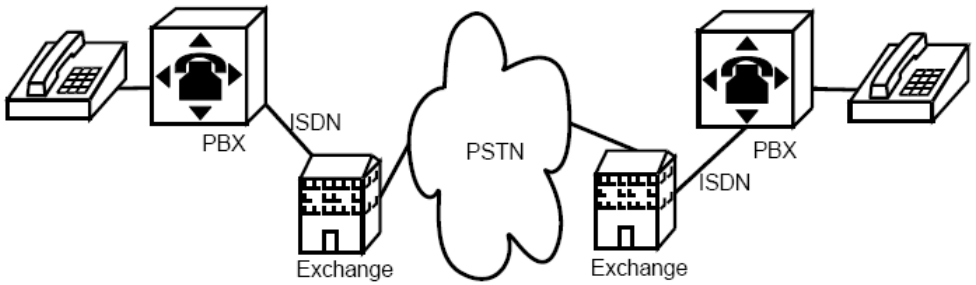
- Digital Signal Processor (DSP) takes with constant interval (*sampling*) the value of the analog signal and associate a binary number (0/1)
- To reduce the throughput (size), the sequence is converted in a compressed format (*CODEC*)
- G.711: flow of 64 Kbit/s, G.729: flow of 8 Kbit/s , GSM : flow of 13 Kbit/s.
- CODEC is a compromised between quality, CPU power, bandwidth, time to transfer (delays), loss tolerance

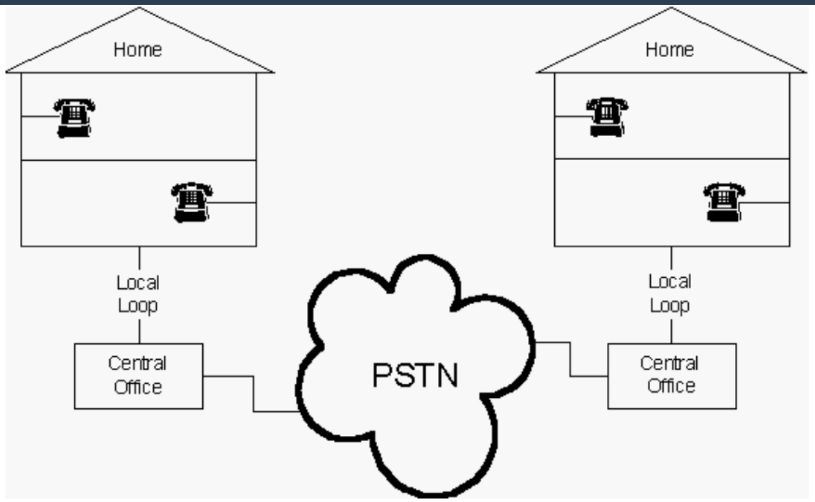


Introduction To VOIP

- What is Public Switched Network ?

  The PSTN is the inter-connection network of all public phone





- Private Branch eXchange (PBX) : Manages calls into and out of organization and does phone number translation
- BRI: Basic Rate, 2 * 64Kbps data channels
- PRI: Primary Rate, 2Mbps (E1)
- Inter-connection between STN network are named TRUNKs
- International dialing plan : Worldwide plan of public numbering: E.164 is an ITU-T recommendation.

Introduction To VOIP

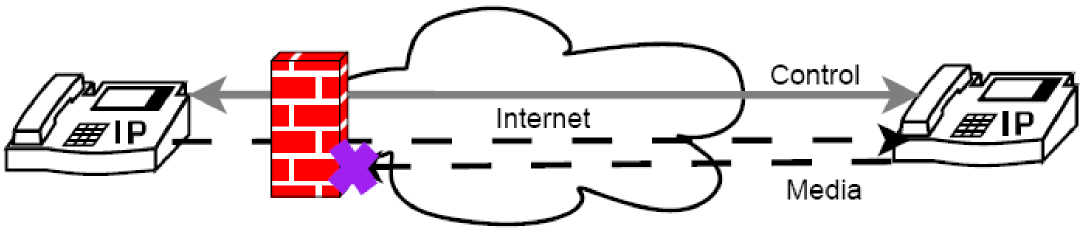
- What is VoIP?

  - Based on packet switching technology using Internet as transport.

  - Opposed to the traditional circuit switching technology, which dominates the Public Switched Telephone Network (PSTN)

  - Driven by low cost.

VOIP is UDP or TCP ?



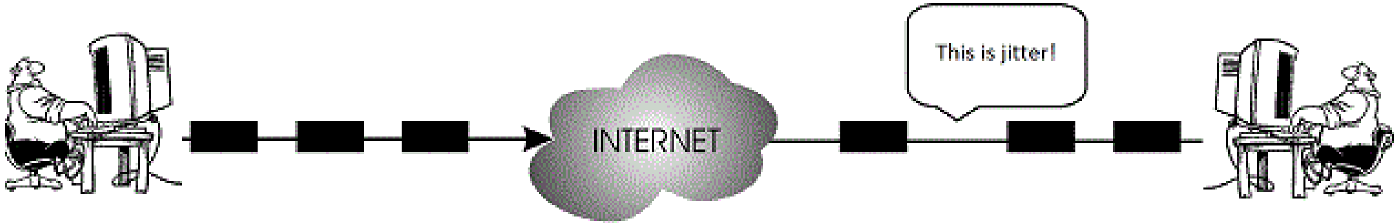
- VoIP: Challenges :

  - Latency

  - Jitter

  - Packet Loss

  - Nat / Firewall Can lead to “one way audio”



What is Asterisk ?

- A complete PBX software for Linux platform developed by Digium
- Asterisk was created by Mark Spencer.
- Does PBX call switching, CODEC translation. and various applications
- Open Source under GNU license

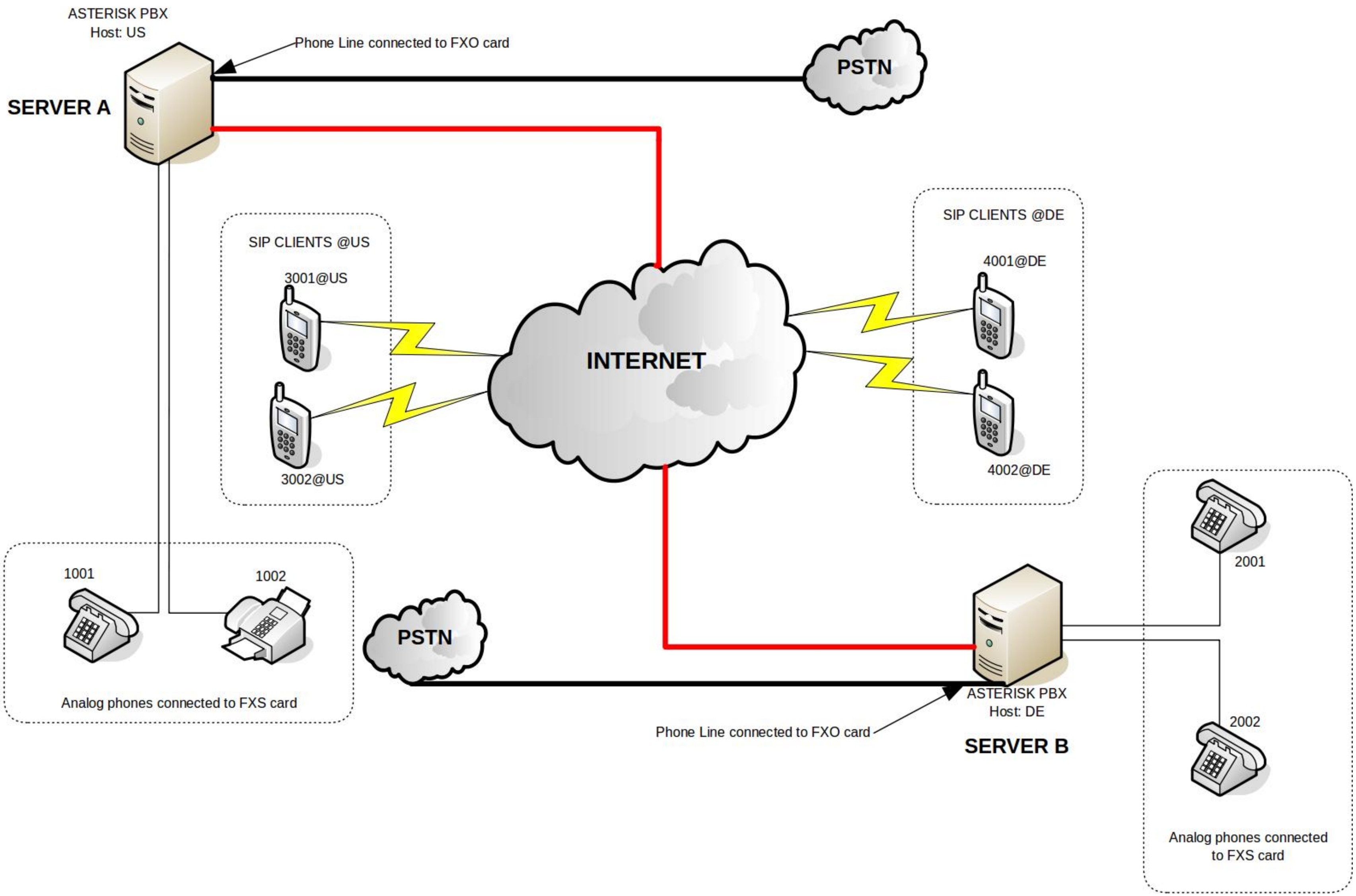
Asterisk Applications

- Voicemail
- Dial an interface (ZAP, SIP, IAX, etc)
- Conference Bridging
- Queues (great for Call centres)
- IVR (press “1” if you know the ext)
- AGI (asterisk gateway interface, like CGI) For advance scripting

Asterisk Configuration files

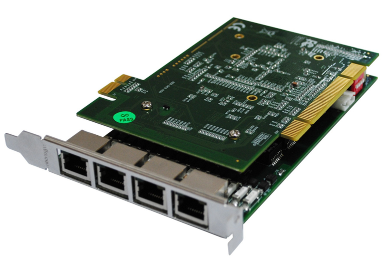
- meetme.conf: Meet Me conference configuration
- musiconhold.conf: Music On Hold configuration
- sip.conf: Configure SIP channels
- iax.conf: Configure IAX channels
- voicemail.conf: VoiceMail configuration
- agents.conf: Configure agent channels
- extconfig.conf: Used by res_data to arrange external configuration
- extensions.conf: The Dialplan

Asterisk Example



VOIP Phones

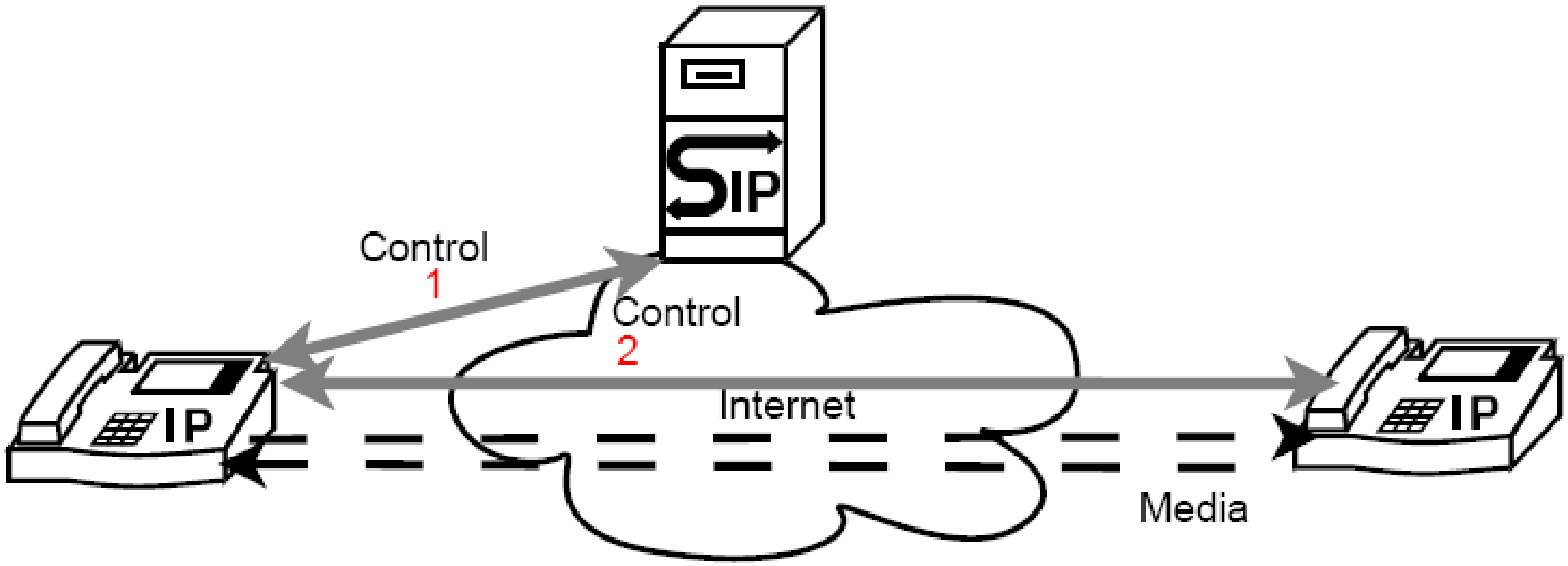
- Analog Phone
- Software Phone
- Hardware Phones (IP Phone)
- TDM POTS /Digital



VOIP Transmission Protocols

- H.323: ITU standard
- SIP: (Session Initiation Protocol) IETF RFC 2543, HTTP-like headers
- SCCP: “Skinny Client Control Protocol”: Cisco proprietary protocol
- Skype: Proprietary protocol based on Kazaa
- IAX2: Inter-Asterisk eXchange v2
- MGCP - Media Gateway Control Protocol

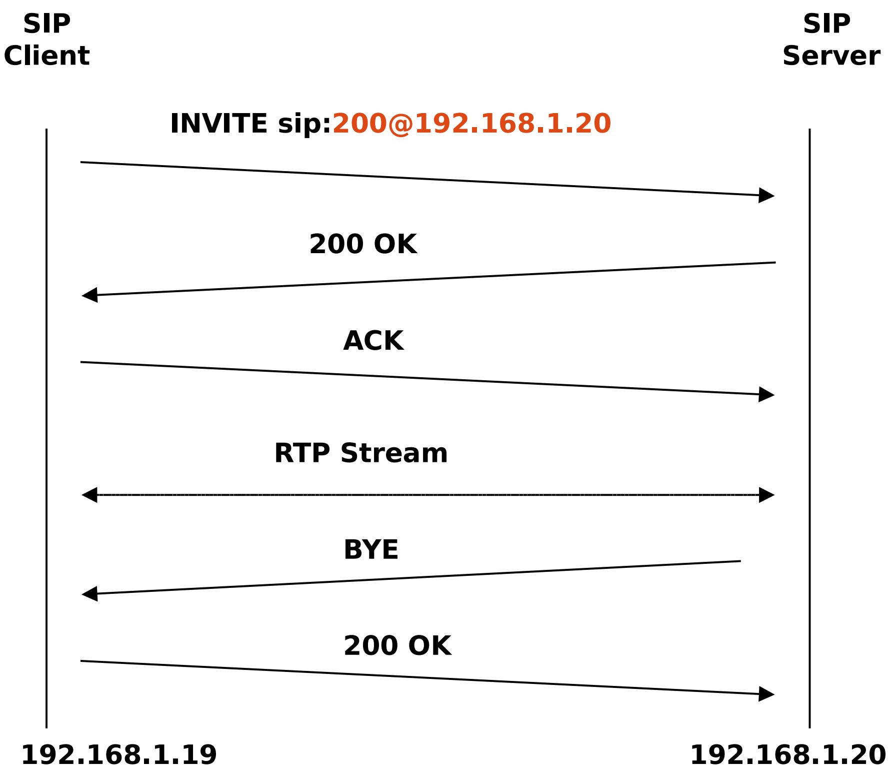
How VOIP Works ?



- All VoIP protocols operate in a similar fashion
- Control channel to set up a call
- Media channels to carry voice data
- RTP: Realtime Transport Protocol , it is ITU standard
- Essentially timestamped UDP packets
- Between dynamically negotiated port numbers
- Similar approach to FTP
- Lots of protocols for control and media channels

SIP Call Setup

Method	Description
REGISTER	Used by client to register a particular address with the SIP server
INVITE	A session is being requested to be setup using a specified media
ACK	Message from client to indicate that a successful response to an INVITE has been received
BYE	A call is being released by either party
CANCEL	Cancels any pending requests. Usually sent to a Proxy Server to cancel searches
OPTIONS	A Query to a server about its capabilities



SIP Responses

	Description	Examples
1xx	Informational – Request received, continuing to process request.	180 Ringing 100 Trying
2xx	Success – Action was successfully received, understood and accepted.	200 OK
3xx	Redirection – Further action needs to be taken in order to complete the request.	300 Multiple Choices 302 Moved Temporarily
4xx	Client Error – Request contains bad syntax or cannot be fulfilled at this server.	401 Unauthorized 408 Request Timeout
5xx	Server Error – Server failed to fulfill an apparently valid request.	503 Service Unavailable 505 Version Not Suported
6xx	Global Failure – Request is invalid at any server.	600 Busy Everywhere 603 Decline

DEMO