Lesson 1

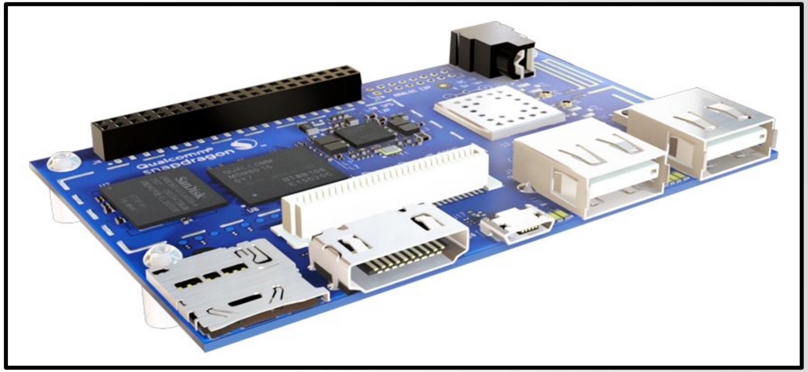
Understanding what VoIP means

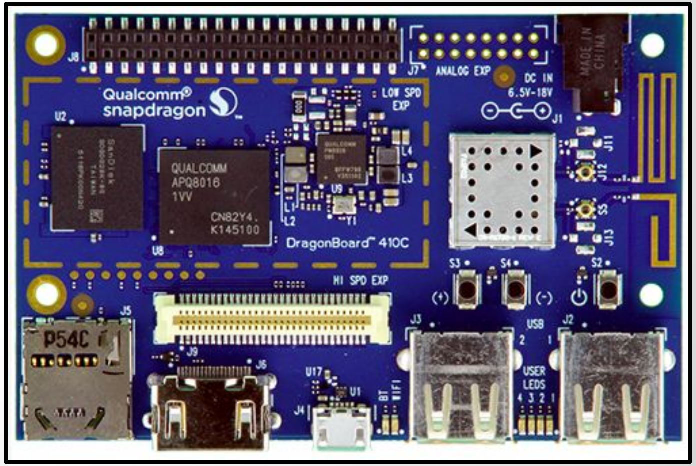
Lesson 1 | Understanding what VoIP means

1 - What is VoIP?

2 - Flavors of VoIP

3 - VoIP: Common applications and Fun Facts





1 - What is VoIP?

1 | What is VoIP?

- VoIP often pronounced “voyp”
- Voice over Internet Protocol
- A phone service over a digital network (The Internet)
- Internet is NOT necessary, Internet protocols ARE

A protocol is a set of rules used to allow orderly communication

- Internet protocols are the basis of IP networking

Supports corporate, private public, cable, and wireless networks

1 | What is VoIP?

- Best and economical way to make international calls
- Easy to setup, easy to use
- Plenty of free services to choose from

<u>Skype, Net2phone, Gizmo, Free World Dialup etc...</u>

- Better voice quality
- Equipped with many features:

<u>Call forwarding, voice mailbox, call records etc...</u>

1 | What is VoIP?

- Analog audio signals ---> Digital data

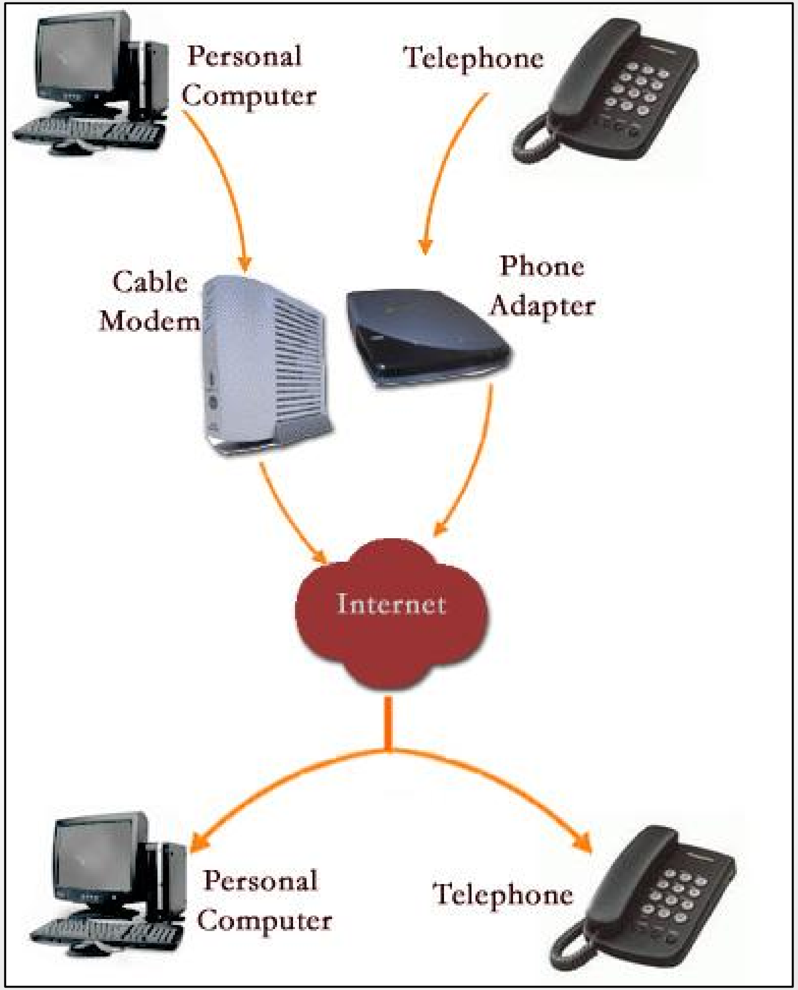
Digital data can be transmitted over the internet

- Three flavors of VoIP

ATA, IP Phones, Computer-to-Computer

- ATA - Analog Telephone Adaptor
- IP Phones - Internet Protocol Phones
- Computer-to-Computer - Computer software based

1 | What is VoIP?



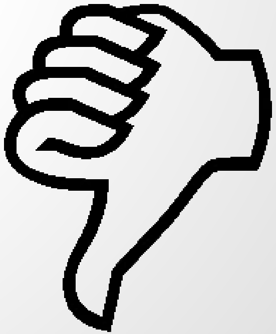
1 | What is VoIP?

Advantages

- Bonus features and services
- Avoid traditional phone costs
- Higher quality

Disadvantages

- No service during power outage
- No emergency services
- No directory services



2 - Flavors of VoIP

2 | Flavors of VoIP

<u>**ATA - Analog Telephone Adaptor**</u>

- Simple and most common form of VoIP

- The ATA is an Analog-to-digital converter

<u>Takes analog from phone and converts to digital for internet transmission</u>

- Allows the connection of a standard phone to computer or internet

- ATAs are bundled with services by providers

<u>Straight forward and easy to set up</u>

2 | Flavors of VoIP

<u>IP Phones - Internet Protocol Phones</u>

- Specialized phone that looks like a normal phone

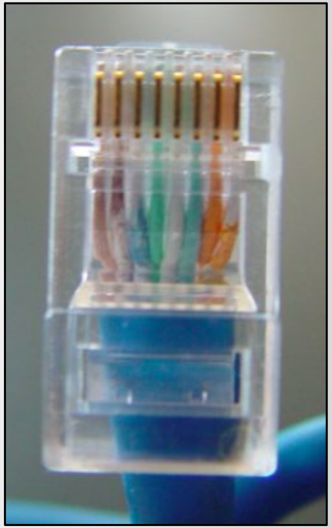
- Standard phone connector replaced with Ethernet connector

<u>IP Phones are connected directly to your router</u>

- All necessary hardware and software is built into the phone

- Wi-Fi phones can make VoIP calls from any Wi-Fi hot spot

<u>Most new Cell Phones have Wi-Fi call capabilities</u>



2 | Flavors of VoIP

Computer-to-Computer

- Easiest way to use VoIP

- In most cases, completely free call to anywhere

This is through free, or very low cost software

- VoIP software, microphone, speakers, internet, sound care and computer

Monthly internet fees will still apply

3 - VoIP: Common Applications and Fun Facts

3 | VoIP: Common Applications and Fun Facts

Computer-to-Computer VoIP - Skype, Teamspeak, Ventrillo, Mumble, etc

Closer Look at Skype:

- Based on peer-to-peer (P2P) networking
- Decentralized and distributed
- "When you sign on to Skype, your computer becomes one node in a global network of equal peers. Each of the clients becomes an active part of the network and, whether it's actively sending messages or not, helps the network as a whole to locate and route traffic to other users."

3 | VoIP**:** Common Applications and Fun Facts

- ***The first VoIP call was made in 1974, on the ARPANET*** - Advanced Research Projects Agency Network, precursor to today's Internet.

- ***Gaming VoIP used by criminals*** - VoIP for gaming is HUGE. Criminals have tapped into this market and are suspected of conducting criminal activities under the disguise of VoIP used for gaming.

- ***Anyone can start their own VoIP service*** - Open source software makes it possible for the experienced coder to make their very own VoIP service

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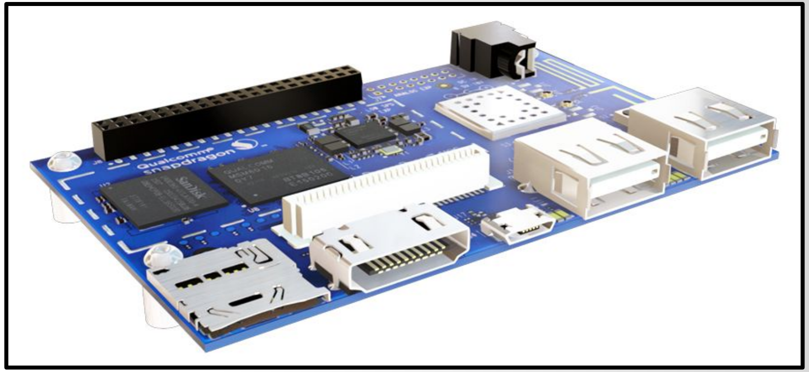
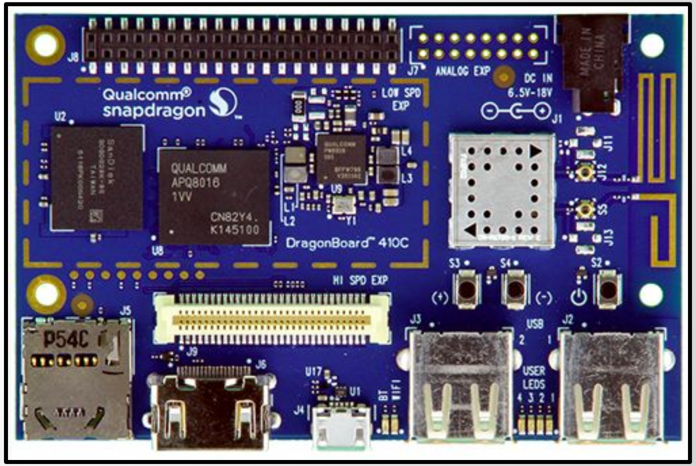
1 - What is VoIP?

2 - Flavors of VoIP

3 - VoIP: Common applications and Fun Facts





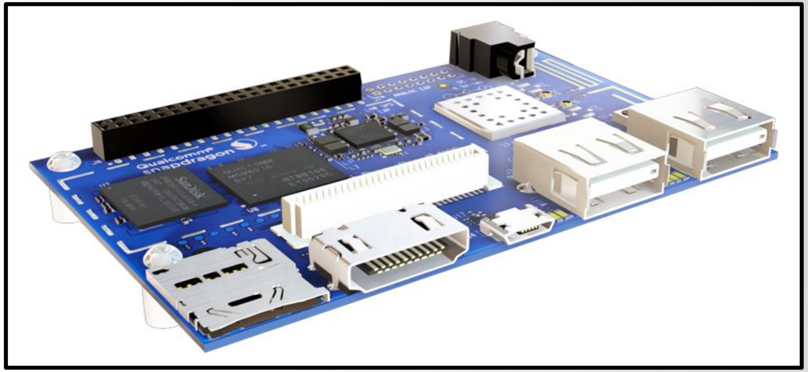
Lesson 2 | VoIP on the DragonBoard™ 410c

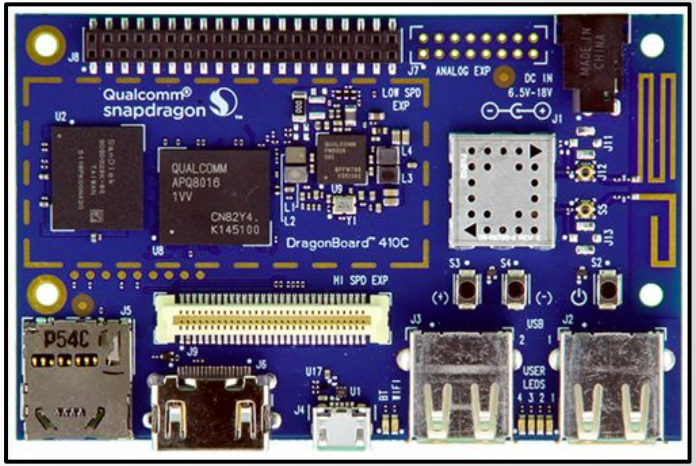
1 - What is Linphone?

2 - Linphone Features

3 - Setup

4 - Linphone Essentials / Walkthrough





1 - What is Linphone?

1 | What is Linphone?

- Open source SIP Phone

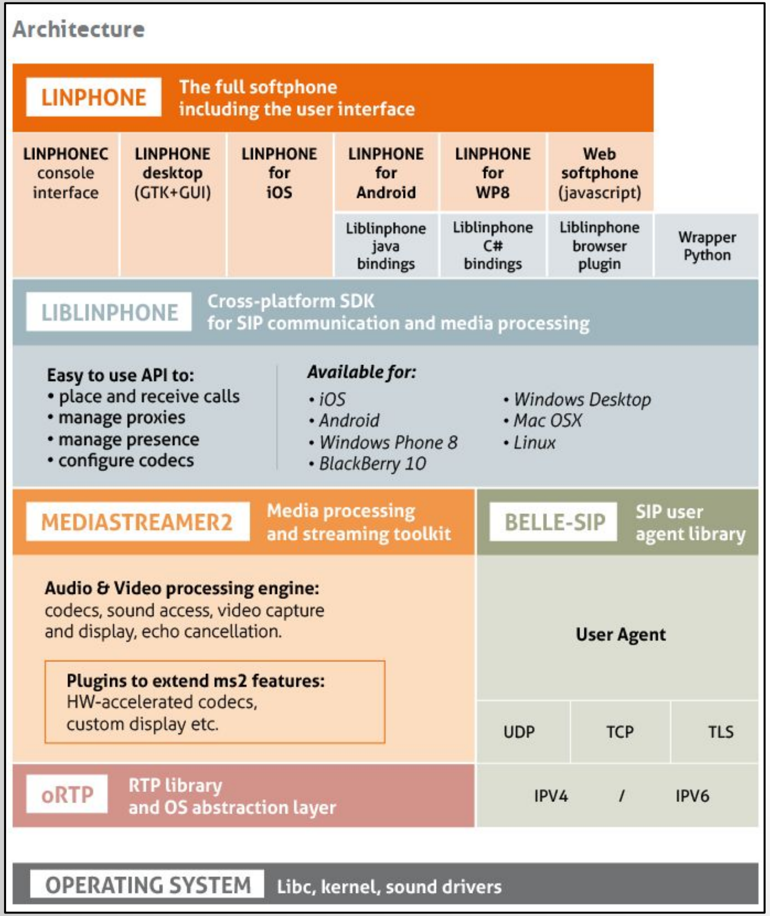
- Available on Mobile and Desktop environments

iOS, Android, Windows Phone 8, Linux, Windows and MAC

- Separation between the user interfaces and the core engine

- Allows the creation of UI on top of core functionalities

1 | What is Linphone?



As seen here:

http://www.linphone.org/technical-corner/linphone/overview

1 | What is Linphone?

Liblinphone:

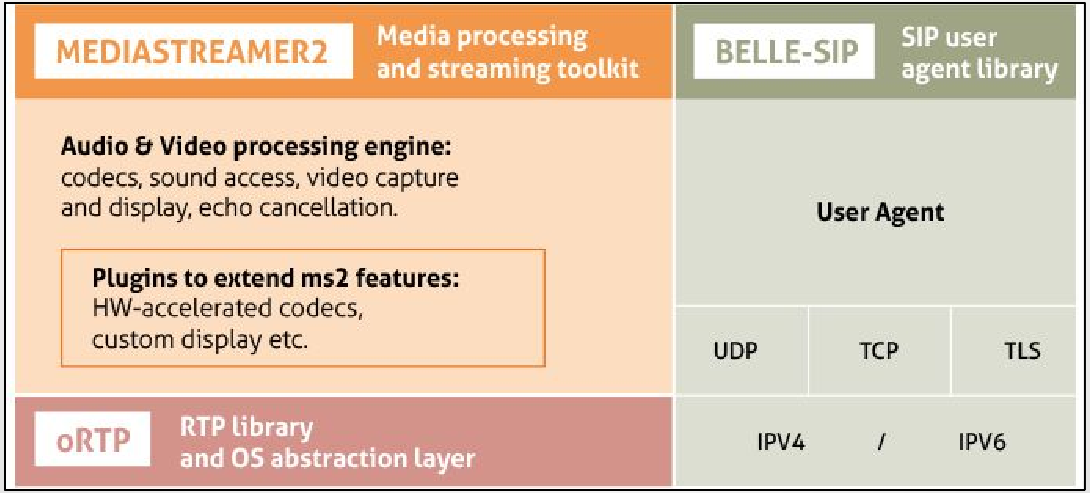
- Core engine
- Implements all functionalities of Linphone
- Powerful SIP VoIP video SDK
- Anyone can add audio or video call capabilities to an application
- Relies on several software components

Mediastreamer2, oRTP, belle-sip

1 | What is Linphone?

<u>**Liblinphone:**</u>

- Mediastreamer2: powerful multimedia SDK used for audio/video processing
- oRTP: simple *Real-time Transport Protocol* library
- belle-sip: Session Initiation Protocol



2 - Linphone Features

2 | Linphone Features

Linphone Features:

- Audio and video calls
- Multiple calls management
- Call transfer, pause and resume
- Audio conferencing
- Instant Messaging

2 | Linphone Features

Linphone Features:

- Pictures and files sharing
- Address Book
- Call History
- Display of advanced call statistics
- Echo Cancellation

2 | Linphone Features

Linphone Features:

- Quality of Service
- Secure communications: zRTP, TLS, SRTP
- Bluetooth headset support
- Multiple spoken languages
- Dedicated tablet user interface

2 | Linphone Features

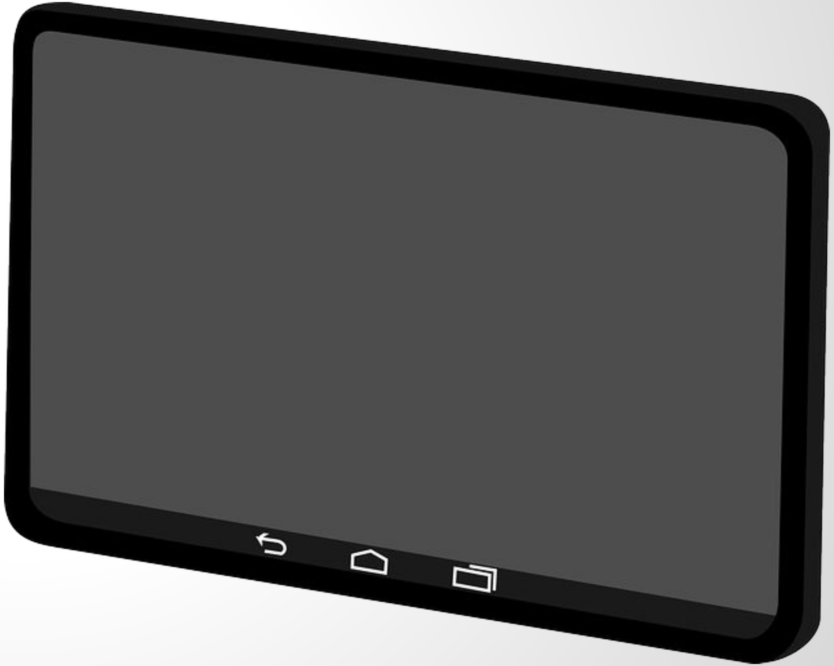
Linphone Features (Advanced):

- Audio Codecs
- Video Codecs
- HD video support
- Push notifications
- ICE support
- Low bandwidth mode

2 | Linphone Features

Applying Features to Your Application:

- Audio calls
- Instant Messaging
- Address book
- Call history
- Profile creation
- File sharing



3 - Setup

3 | Setup (Download)

Download Steps (Android):

- Go to www.linphone.org
- Under “Technical Corner” click LINPHONE
- Click DOWNLOADS tab at top of page
- Scroll down to Linphone Android on Google Play and the apk here
- Click here for apk download

3 | Setup (Download)

Download Steps (Ubuntu):

- ***Launch on board terminal and execute:***

$ sudo apt-get update

$ sudo apt-get upgrade

$ sudo apt-get install linphone

3 | Setup (Registration)

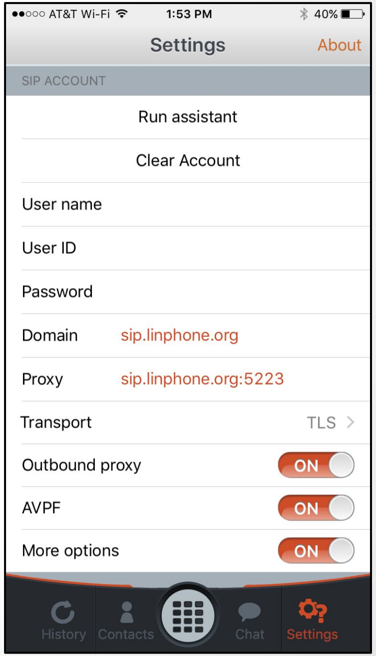
- ***Go to*** - www.linphone.org
- Click CREATE OR MANAGE YOUR ACCOUNT > in the “Free SIP Service” box
- Fill out information under: Create an account
- Click CREATE NOW ! and you are ready to go
- ***Network*** - Server, ports and media encryption
- ***Advanced*** - Debug, and more server options

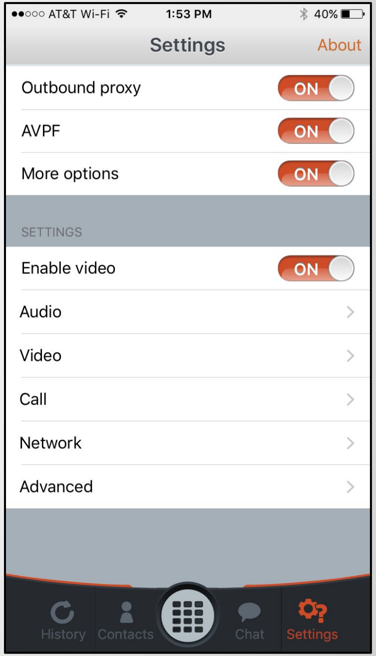
4 - Linphone Essentials

4 | Linphone Essentials / Walkthrough

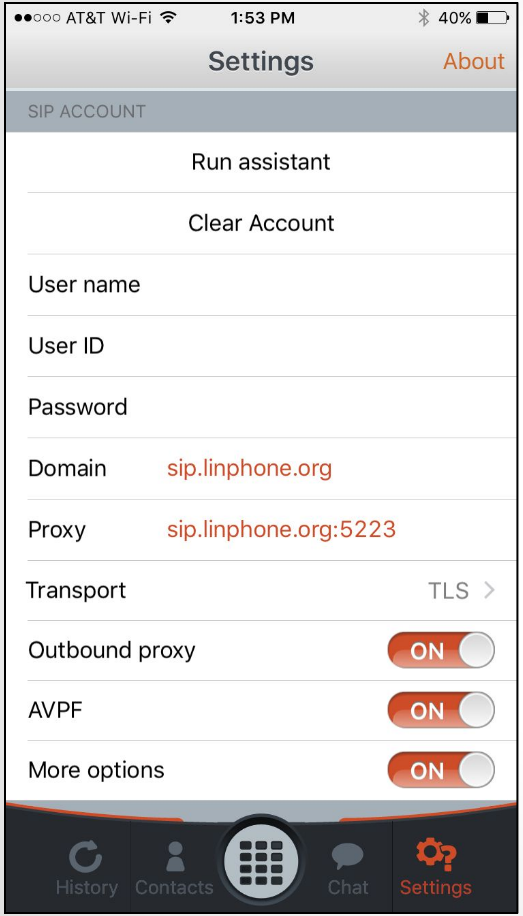
Settings:

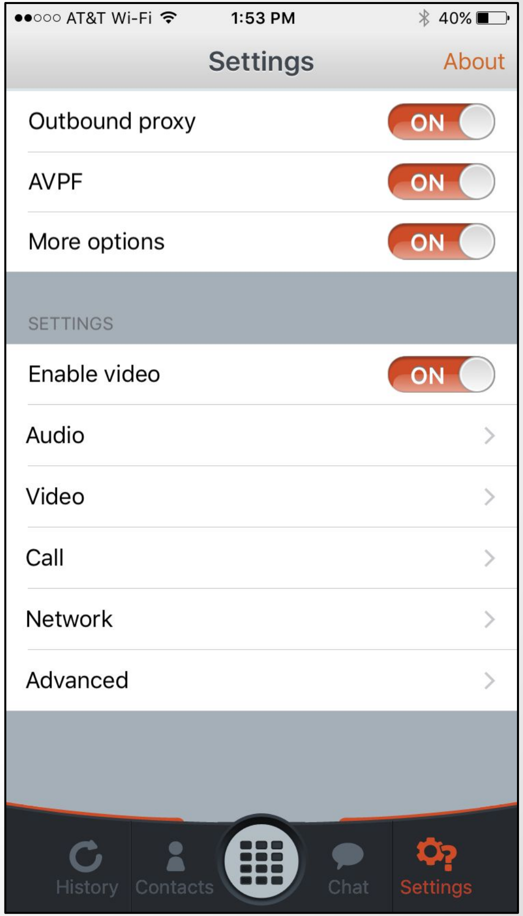
- ***SIP Account*** - Identity and domain
- ***Audio*** - Codecs and bitrate
- ***Video*** - Codecs
- ***Call*** - Prefix, send and sub options
- ***Network*** - Server, ports and media encryption
- ***Advanced*** - Debug, and more server options





4 | Linphone Essentials / Walkthrough





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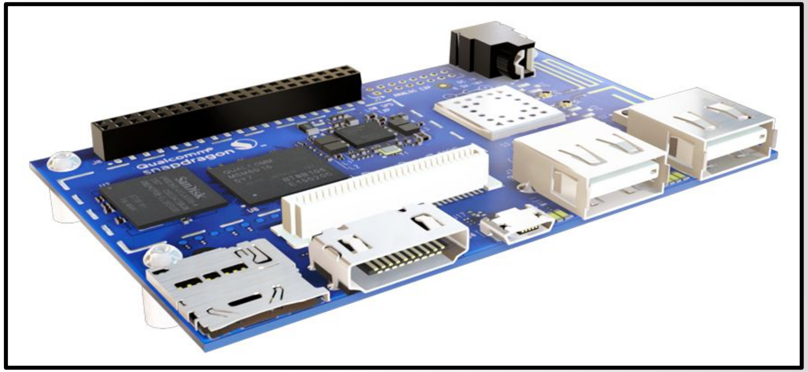

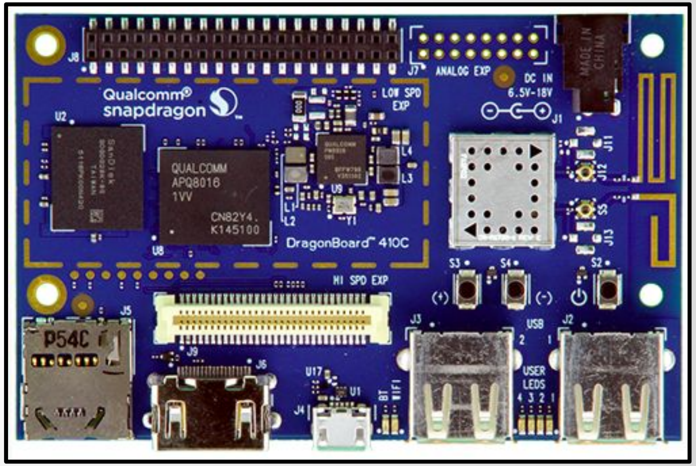

Lesson 3

Closer Look at VoIP and SIP

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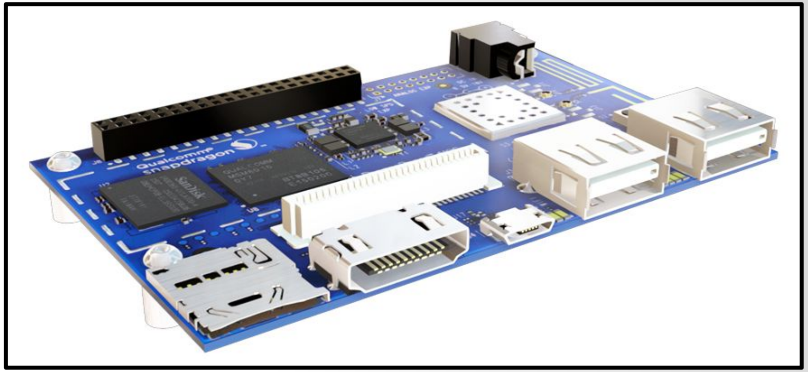
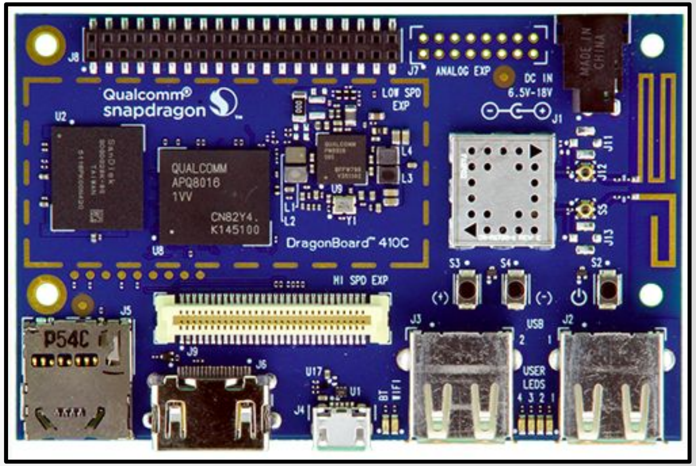




1 - SIP

2 - Protocol Operation

3 - Network Elements

4 - SIP Messages

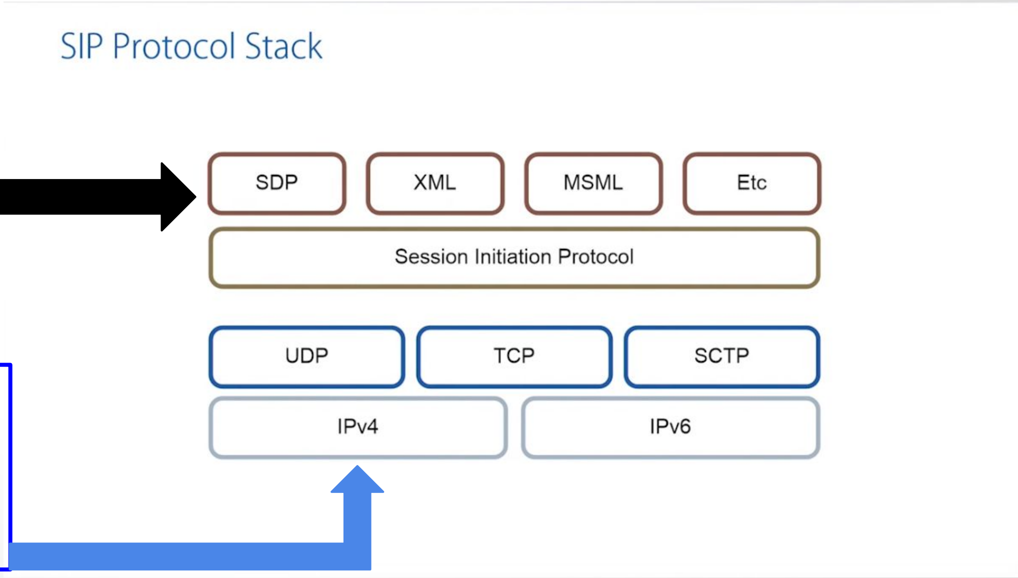
1 - SIP

1 | SIP

- stands for **Session Initiation Protocol**
- a communications protocol for signaling and controlling multimedia communication sessions.

- It can carry a variety of payloads

- It can be carried **over** a variety of different transport protocols

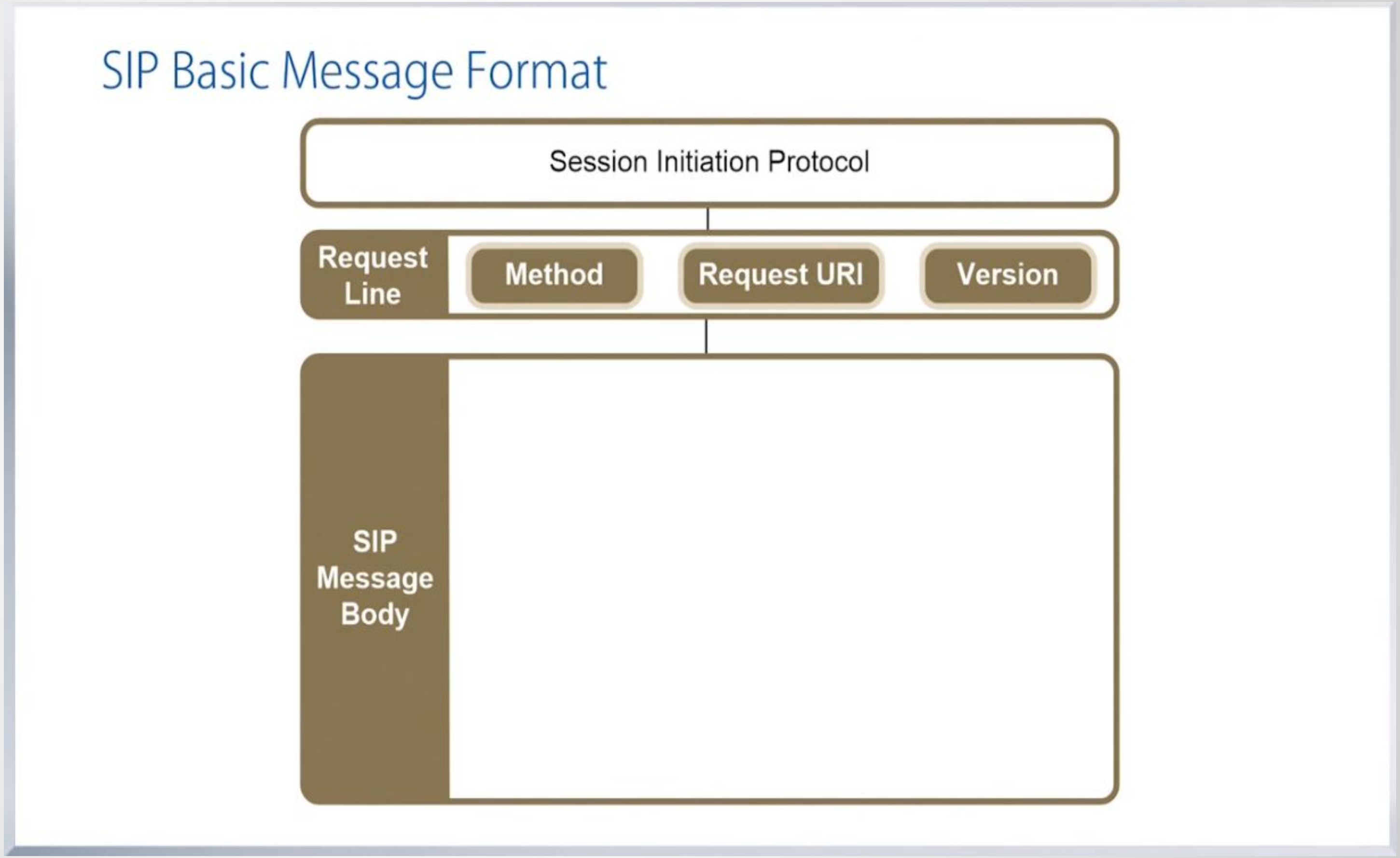


1 | SIP

- Message - What is the SIP doing? What is the point of the request?

- Request URI - Who is this request actually going out to?

- Version - Version of the operation



2 - Protocol Operations

2 | Protocol Operations

- Runs on TCP, UDP or SCTP (two-party or multi-part)
- Similar design elements as HTTP request/response model
- Client request invokes method or function and a response
- Reuses HTTP header files, encoding rules and status codes
- SIP network resources are identified using a URI

URI: Uniform resource identifier

- Typical SIP URI: *sip:username:password@host:port*

2 | Protocol Operations

- SIP is only involved in signaling portion of communication
- Clients typically use TCP or UDP on ports 5060 or 5061
- Port 5060 used for non-encrypted signaling traffic
- Port 5061 used for traffic encrypted with Transport Layer Security (TLS)
- Used for setting up and tearing down voice and/or video calls
- Voice and video stream carried over by RTP

3 - Network Elements

3 | Network Elements

- User Agent
- Proxy server
- Registrar
- Redirect server
- Session border controller
- Gateway

3 | Network Elements

<u>**User Agent (UA):**</u>

- Logical network endpoint, creates and receives SIP messages

- Manages a SIP session

- UA ---> UAC: sends SIP requests; UAS returns SIP response

These roles last for duration of SIP transaction

- A SIP phone is an IP Phone that implements UAS functions!

Essentially providing the functions of a telephone

- Implemented as a HW or SW SIP device

3 | Network Elements

Proxy Server:

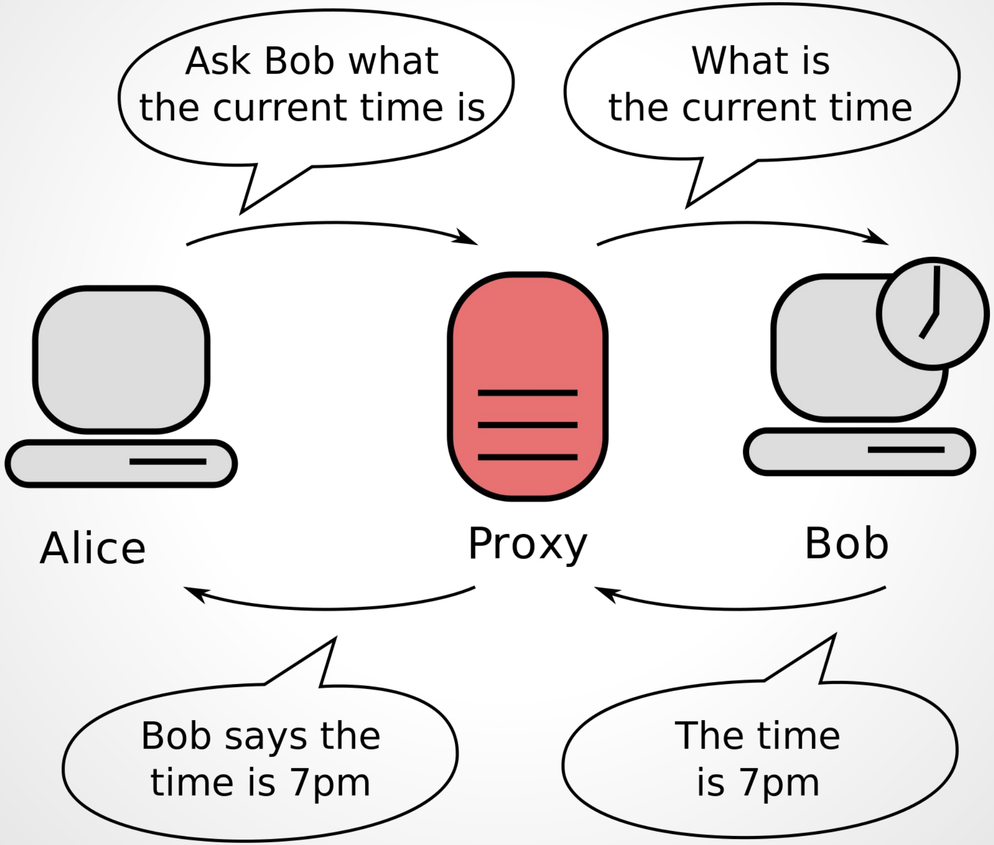
- Intermediary Entity, acts as as both server and client
- Performs requests on behalf of other clients
- A proxy server primarily plays the role of routing

Ensure a request is sent to another entity closer to target user

- Proxies enforce policy, call permissions

3 | Network Elements

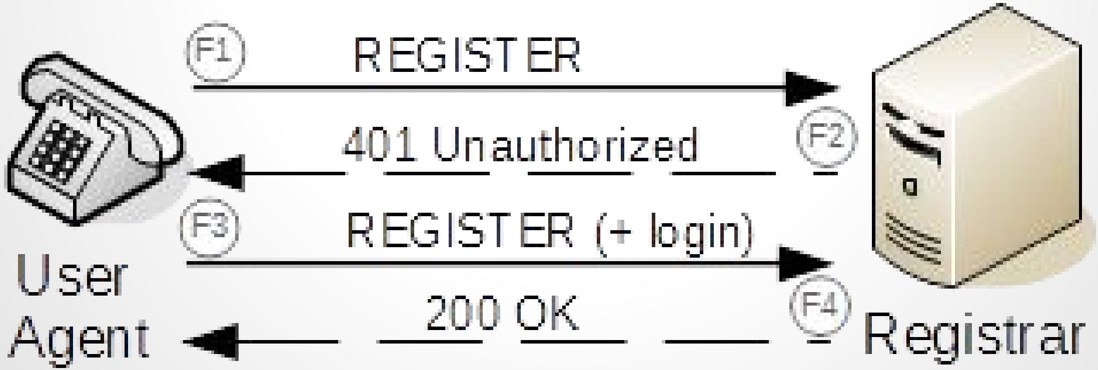
Proxy Server:



3 | Network Elements

Registrar:

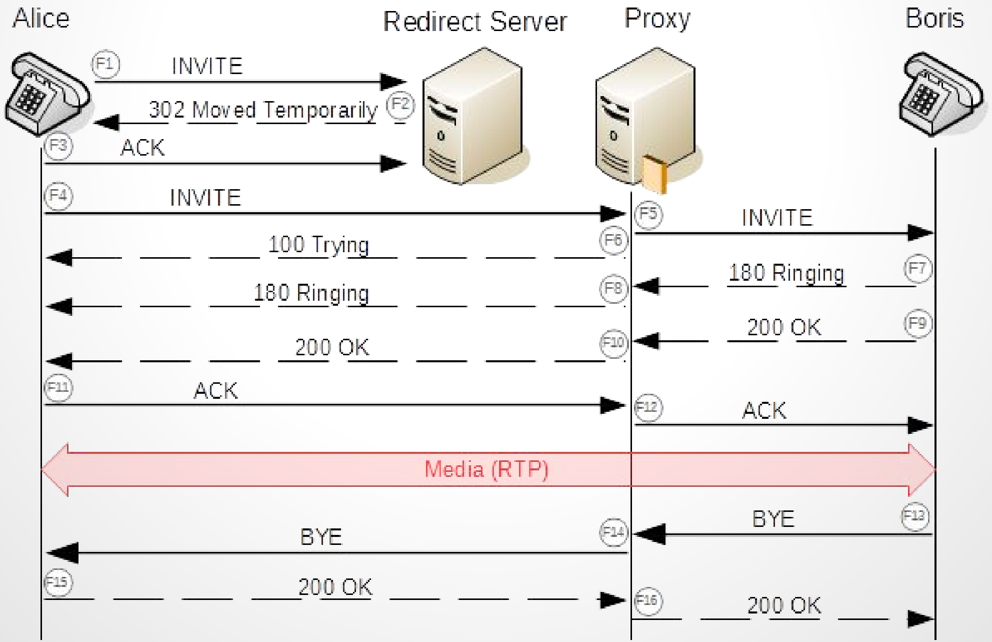
- SIP endpoint, accepts REGISTER requests
- Information received in requests, placed into location service
- Loc service links IP addresses to SIP URI and registering agent



3 | Network Elements

Redirect server:

- Redirect server allows proxy servers to direct SIP sessions invitations to external domains



3 | Network Elements

Session border controller:

- Exerts control over VoIP signaling, setup, during, tear down.

Gateway:

- Used to interface SIP network with other networks, such as the PSTN

4 - SIP Messages

4 | SIP Messages

<u>SIP request</u>

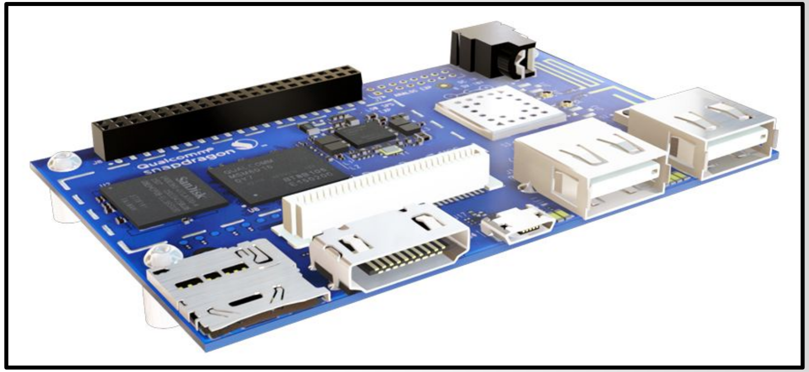
- **REGISTER:** Used by UA to register to the registrar
- **INVITE:** Used to establish a media session between UAs
- **ACK:** Confirms reliable message exchanges
- **BYE:** Terminates an existing session
- **CANCEL:** Terminates a pending request
- **OPTIONS:** Request info about capabilities of caller without session
- **REFER:** Indicate recipient should contact third party w/ provided info
- **PRACK:** Improves network reliability, adds acknowledgement system to provisional responses

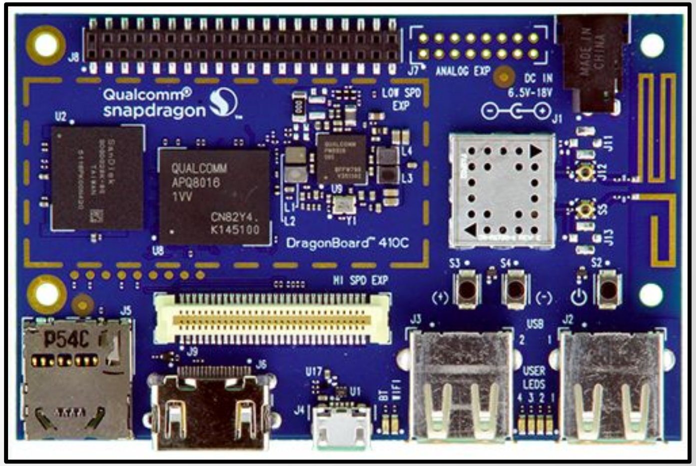
4 | SIP Messages

SIP response

- **Provisional:** Request received and being processed
- **Success:** The action was successfully received, understood and accepted
- **Redirection:** Further action needs to be taken to complete the request
- **Client Error:** The request contains bad syntax or cannot be fulfilled at the server
- **Server Error:** The server failed an apparently valid request
- **Global Failure:** The request cannot be fulfilled at any server

Lesson 3 | Summary + A Look Back





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