# Assignment-3
# Dial-a-Song

# Assignment Motivation

- **Understanding SIP protocol (in a practical way)**
  - Questions like: why certain headers are needed, purpose etc?
- **Working with real world devices**
  - Writing program that interact with real world devices
- **If you have to design a protocol in future**
  - it helps to have practical experience of working (code) with protocols

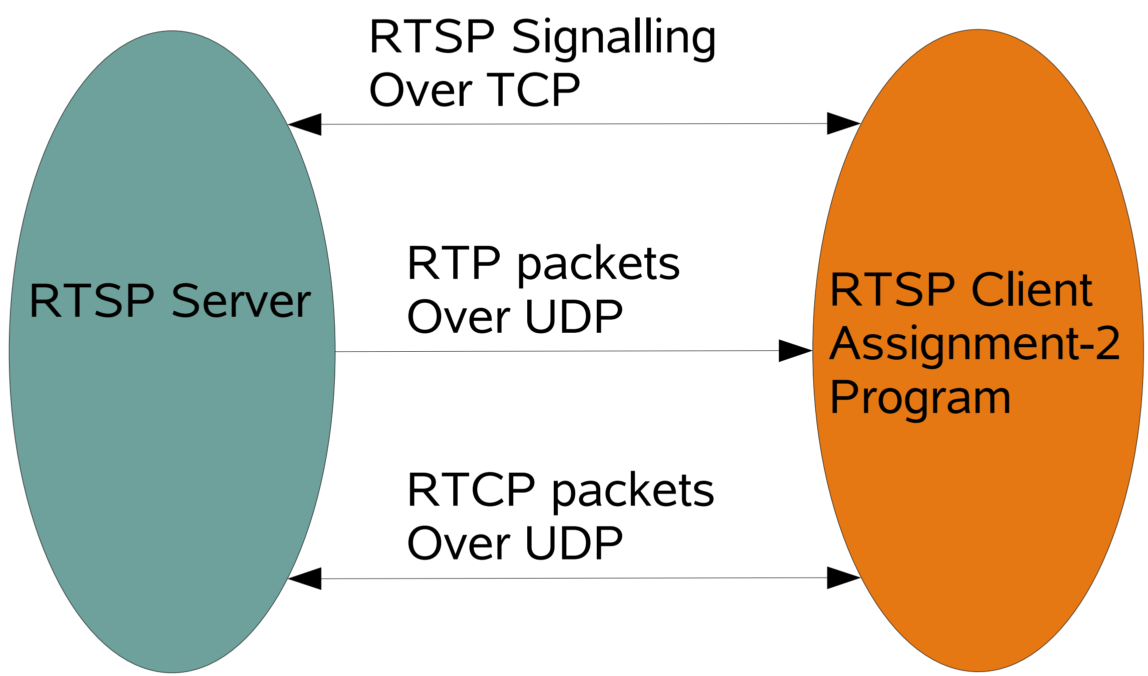
# Assignment Overview

- A SIP Client calls a specific SIP user-id and gets a music stream to its terminal
- Can be considered as a **Music-On-Demand** service (assignment does not require video support, but if supported could also be considered a Video-On-Demand Service)
- Using SIP for On-Demand service seems an interesting idea.
- Components Involved:
  - A SIP softphone
    - Example: kphone, X-lite - Can be freely downloaded
  - A RTSP server (already UP and running at Netlab)
    - rtsp://130.233.154.184:8554/song1.wav
  - The assignment program that interacts with both the above components

# Assignment Details

- Must be able to handle requests from a chosen SIP client
  - Your program need to pose as a SIP server
    - Building a complete SIP server is **not** the task (neither possible in the given time)
    - Need to have functionality to successfully establish a voice call
      - i.e. functionality to process and respond REGISTER, INVITE, ACK and BYE messages
  - SIP Clients support both signalling over TCP and UDP
    - Choose one, that you prefer to implement (TCP or UDP)
- Interface with the Assignment-2
  - Getting the media stream from the RTSP server. (would use the Assignment-2 RTSP Client functionality)
  - But the media stream destination need to be specified as SIP client

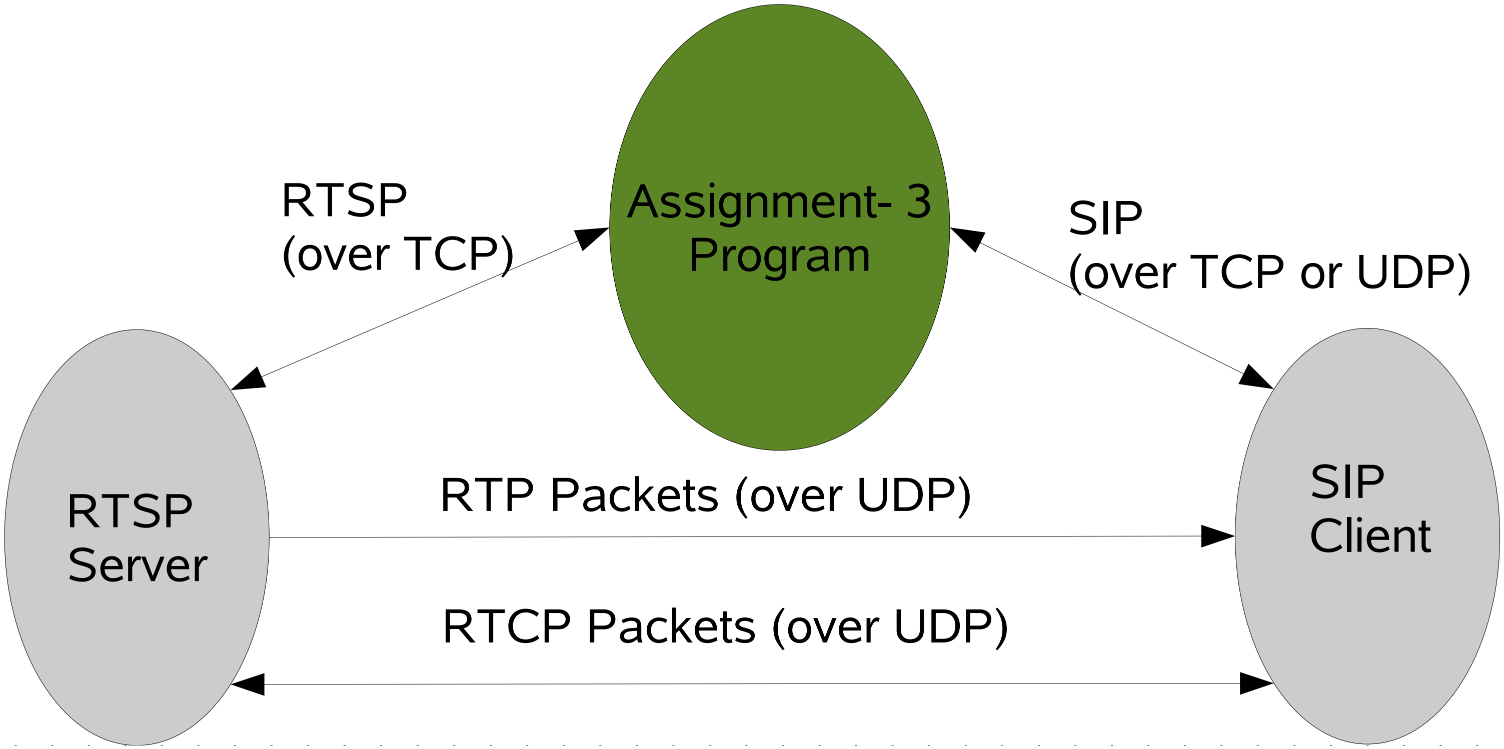
# Recap: Assignment-2 Overview




**Hints:**

1. DESCRIBE Response contains SDP description
(SDP parameters carry media format details)
2. SETUP Request and Response carries address parameters

# Assignment-3 Overview




- In Assignment-2, the media data was received by RTSP client.
- But in Assignment-3, the media data destination need to be redirected

**Hints:** Media format in SIP response need to be based on what RTSP server provides

# Program Execution Flow

- SIP client calls your Assignment-3 program.
  - SIP user-id ***sip://song1@address.com***
- Receive the call
  - Extracts the user-id. (song1)
- Initiates RTSP session with the media server (but the media destination need to be modified)
  - *rtsp://130.233.154.184:8554/song1.wav*
- After receiving PLAY response from RTSP Server, send 200 OK to the SIP client.
- Now the media is played at the SIP Client
- Also, take care of ending the session cleanly

# SIP clients

- X-Lite
  - http://www.counterpath.net/x-lite.html
  - OS support: Windows, Linux and Mac
- kphone
  - Download from http://www.wirlab.net/kphone/kphone-4.2.tar.gz
  - Source code available

There are many other clients available, you are free to choose your SIP client.

# Submission guidelines

(for both Assignment 2 and 3)

- All required source files
- A readMe file with details on compilation and execution instructions
- Brief comment about your assignment (approx. 1page)
  - Implementation issues faced
  - Comments/Suggestions/Complaints, if any
  - Extra features if any
  - Anything that you would to like to tell us
- There shall be a demo of the submitted assignments
  - Demo date, yet to be finalized.
  - Follow announcements in Noppa
  - Deadline: One Final Deadline for both Assignment 2 and 3
    - January 9th 2009 (No extension possible, So start early)