RTSP to WebRTC Gateway

A start-up in security cameras needed an Android Application to view live camera feeds with ability to stream the content to web browser using WebRTC.

Project Size: 5 members
Project Duration: 7 months
Technology Used: Android, NDK, RTSP, WebRTC, H.264, AAC, Opus, transcoding, Socket programming
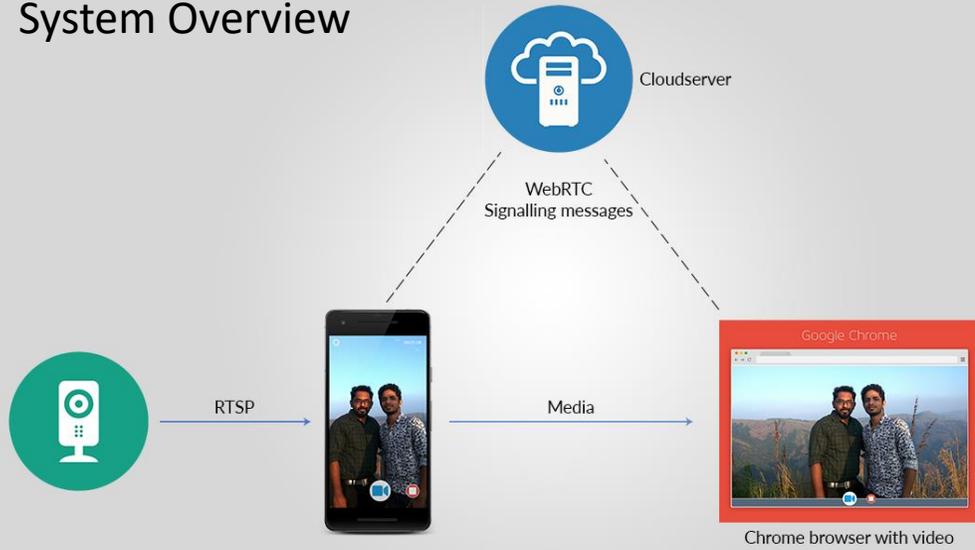
Challenge

- Implementing a mechanism to forward video to WebRTC without need for transcoding.
- Transcoding AAC audio from camera to Opus for WebRTC as WebRTC does not support AAC
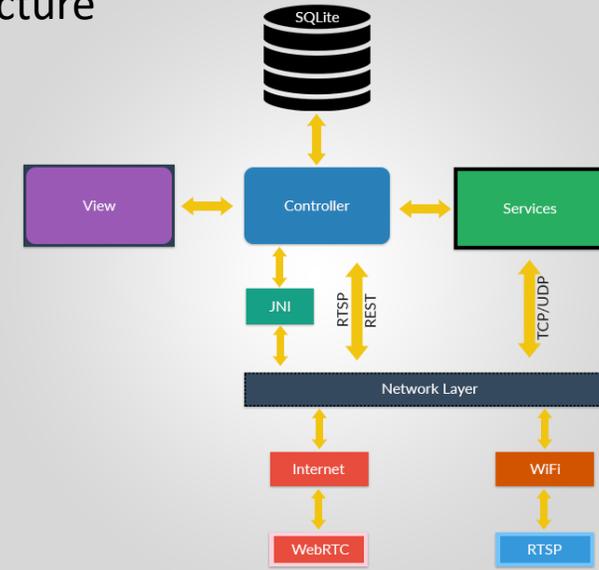
Outcome

- Developed Android application with support for RTSP and WebRTC
- Support local playback of video, and capturing images and videos
- Developed WebRTC server application with user interface to list active cameras and view the stream
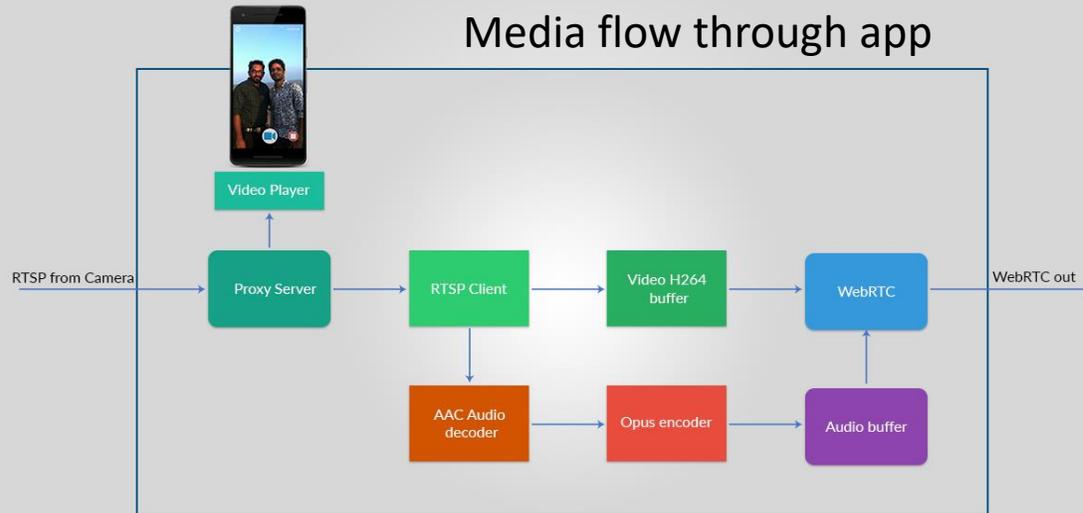
System Overview

Cloudserver

WebRTC
Signalling messages

RTSP

Media

Google Chrome

Chrome browser with video

App Architecture

SQLite

View

Controller

Services

JNI

RTSP
REST

TCP/UDP

Network Layer

Internet

WiFi

WebRTC

RTSP

Media flow through app

Video Player

RTSP from Camera

Proxy Server

RTSP Client

Video H264
buffer

WebRTC

WebRTC out

AAC Audio
decoder

Opus encoder

Audio buffer

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WHAT DID GADGEON DO?	PLATFORMS/ TECHNOLOGIES USED
1) Software Architecture for complete system	Requirements Gathering & Analysis, Uses cases, Architecture development, identification of functional components.
2) Android application development	Discover camera by listening to UDP beacons and listing them. Connect to the user selected camera. Display the live video stream on the app UI. Configure the camera through WiFi APIs. Various options like Resolution, Video quality, Photo size, Photo quality etc can be configured. Option to download images and videos from Camera to the Android Gallery. Option to record the media locally from the incoming RTSP stream. Generate and display media thumbnail for videos and photos. Implement Audio only mode. Implement live streaming to WebRTC on user tap. Continues to play video on the screen. Streaming continues even when app is in background. App implements H.264 frame forwarding from RTSP to WebRTC. Audio is transcoded from AAC to Opus.
3) Cloud application development	Google Cloud platform is used for WebRTC Web server and for Signalling server. Implemented Web socket based signalling server. Used STUN/TURN servers for NAT/Firewall Traversal. WebRTC Web server has the following features:- • UI page to view the live stream from camera • UI page to list all the active live streams from different cameras • Provision to record the live stream

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