Signal Processing System for Imbedded Transparent Ultrasonic Emitter in Smartphones

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1. Summary

Applicable transparent parametric loudspeaker for smartphones has become available recently, yet few reports have mentioned the detailed design of signal processing system before the emitting stage. The purpose of this project is to develop such a system which pertains to the emitter and exhibits satisfactory characteristics to meet the industry need. Specifically, the system can help provide high-quality audio in an efficient and economic manner, be easily imbedded into a smartphone and function well in harsh conditions with a long lifespan. A prototypical implementation of the system comprises of six modules: an equalizer to handle flat response, a compressor to narrow the range of audio amplitude, a low pass filter and a high pass filter, a modulator to combine the signal with a carrier ultrasonic wave generated from the oscillator and another high pass filter to eliminate all audio frequencies. Design considerations of each module are presented in great detail in this proposal. A description of operational elements, including proposed timetable and personnel, budget estimation and evaluation procedure are provided as reference. The ultimate product of this project is a well-performing signal processing system for imbedded transparent parametric loudspeaker in smartphones, which is expected to both fill the gap of the missing design and bring considerable economic benefits.

2. Introduction

The fundamental theory of parametric loudspeaker is raised by Westervelt as ”parametric end-fire array” (Westervelt, 1960), explaining that when two plane waves propagate in the same direction, they will generate some new waves, one of which has a frequency equal to the difference of the two original frequency (see FIG. 1). Several equations describe the model in various dimensions (Westervelt, 1963; Zabolotskaya, Khokhlov, 1969; Kuznetsov, 1971; Berktay, 1965) and the new parametric loudspeaker, or namely “audio spotlight” is developed based on the farfield array condition of these equations (Yoneyama et al., 1983). Progress in these field mainly focus on parameter optimization (Moffett and Mellen,1977; Aoki et al., 1994), distortion reduction (Kamakura et al., 1984; Kite et al., 1998), attenuating large amplitude of ultrasound (Kamakura et al., 1984), propagation model of finite amplitude sound beams (Yang et al., 2005) and so on.

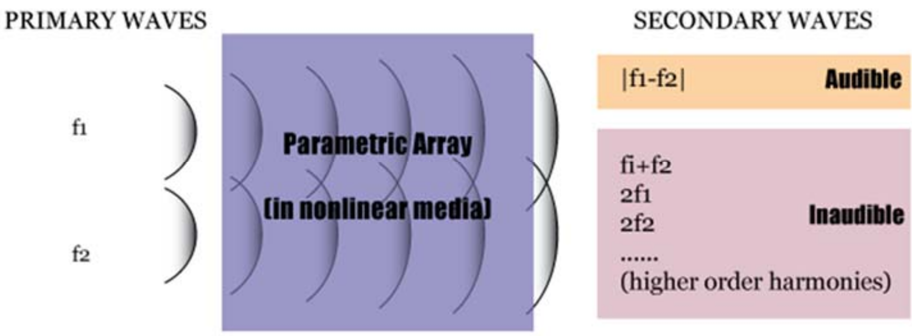

Figure 1: Nonlinear interaction process

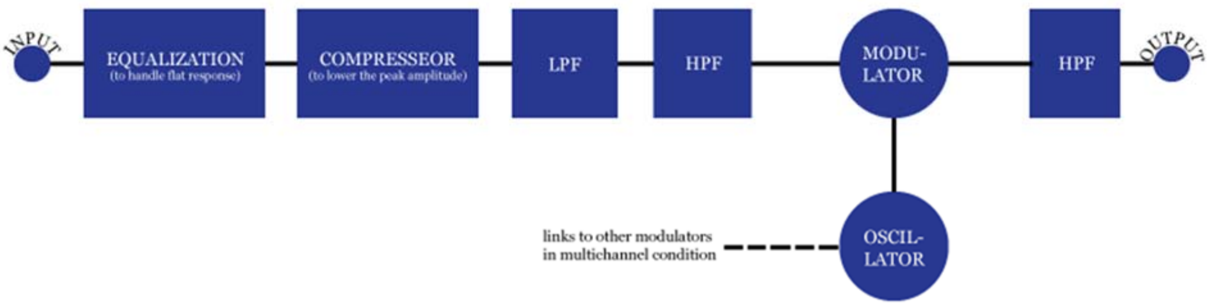

Figure 2: The model of parametric loudspeaker

Some practical implements of parametric loudspeaker have been developed to meet the industrial need. One of them uses two conductive membranes of high transparency (Hecht et al., 2016) with an insulating air gap between the layers, where generation of ultrasound is due to the electrostatic force on the capacitor (Haller and Khuri-Yakub, 1996). It allows an application of parametric loudspeaker on display screen, e.g. screen of smart phone or television, which has a wide range of commercial potential in current market. However, few reports have mentioned the detailed design of signal processing system before the emitting stage.

The goal of this study is to develop a practical signal processing system that can well performed for the transparent parametric loudspeaker. This system should also be solid, extendible and well performed, specifically handle the problem of (1) finding optimal parameters, (2) distortion reduction and (3) attenuating large amplitude of ultrasound (Kamakura et al., 1984). The prototyping of this system consists of six main components (see FIG. 2): an equalizer to handle flat response, a compressor to narrow the range of audio amplitude, a low pass filter and a high pass filter, a modulator to combine the signal with a carrier ultrasonic wave generated from the oscillator and another high pass filter to eliminate all audio frequencies. Design or improvement of the ultrasound emitter is excluded from our current purpose.

3. Objectives

Our signal processing system is critical to the actual performance of the transparent ultrasonic emitter, as well as successful incorporation into a smartphone. Taking these aspects into account, our desired system should exhibit the following characteristics:

- High audio quality. Our system should be able to compensate distortions introduced in audio transmitting and emitting, creating an outstanding experience for users.
- Miniaturization. In order to imbed our system into a smartphone (e.g., iPhone7, measuring 5.44×2.64×0.28 inches), its size and weight should be minimized.
- Low power consumption. Since most smartphones are powered by a single battery, our system should not sever as a further burden. Its power should be consummate, if not negligible compared to other parts of a smartphone.
- Harsh condition resistance. Our system should be operative under similar conditions a normal smartphone can sustain, if not worse. For example, it should function appropriately at a temperature range of 0~40°C.
- Robust. The lifespan of our system should be over the average lifespan of smartphones at present. Four years may be an upper limit for that.
- Economic. The price of our system and the ultrasonic emitter together, when in mass production, should be acceptable for smartphone producers.

4. Procedures/Scope of Work

As discussed above, we are going to design an audio signal processing system to convert the origin audio signal into our expected ultrasonic level signal. The brief diagram of our proposed system is shown as FIG2, as suggested by the author of the emitter (Hecht et al., 2016.). The diagram in FIG. 2 is a single channel system, while it can be extended to a multi-channel system easily by adding repeated structure.

First, we use equalizers to adjust the balance between frequency components within an electronic signal, that is, to strengthen (boost) or weaken (cut) the energy of specific frequency bands or frequency ranges. Then, compressors are introduced to compress the dynamic range of the incoming signal, effectively raising or lowering the amplitude of some certain components of signal. After that, we introduce low-pass filters (LPFs) and high-pass filters (HPFs) to filter out the useless frequency components, which are either out of human hearing range or may compose some unexpected audible sound. Then we modulate the preprocessed signal with the carrier signal to generate an ultrasonic audio signal. Finally, HPFs are applied again to eliminate the high frequency noise, which may be introduced by modulation.

Equalizer

Parametric equalizer (or parametric "EQ") is an electronic multi-band variable equalizer used in sound recording and live sound reinforcement. They allow audio engineers to control the three primary parameters of an internal band-pass filter: amplitude, center frequency and bandwidth. Bandwidth is typically represented by Q, the quality factor. The amplitude of each band can be controlled, and the center frequency can be shifted, and widened or narrowed.

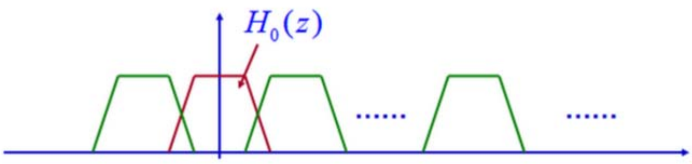

Figure 3: Schematic diagram

In our design, parameters of EQ are determined by the inherent characteristic of the ultrasonic emitter in the following manner. Firstly, inherent attenuation and amplification of different frequency components are detected. Then, the equalizer is tuned to boost or cut respective frequency components, compensating the distortion.

To measure the ultrasonic emitter inherent characteristic, we detect the amplitude of the signals on different frequency under the condition of a specified power. If the detected frequency spectrum curve beyond the scope of international standard, we change parameters for a compensation until the curve in the range.

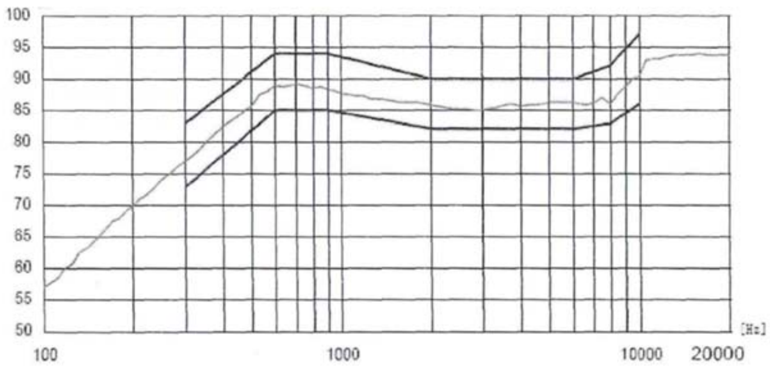

Figure 4: International standards 1

Parameters calculation is based on the requirement of international standard.

频率	上限 SPL（dB）	下限 SPL（dB）
300	83	73
600	94	85
900	94	85
2000	90	82
6000	90	82
8000	92	83
10000	97	86

Figure 5: International standards 2

The method for the equalization of audio systems is using IIR (infinite impulse response) parametric filters. Four-order Butterworth high-pass filter tansfer-function can be written as:

$$H(s) = \frac{(s/\omega_1)^4}{(s/\omega_1)^4 + 2.6131(s/\omega_1)^3 + 3.4141(s/\omega_1)^2 + 2.6131(s/\omega_1) + 1}$$

ω_1 is the normalized angular frequency, the function can be regard as the multiply of two seperated function:

$$H(s) = H_1(s)\times H_2(s)$$

$$H_1(s) = \frac{(s/\omega_1)^2}{(s/\omega_1)^2 + 1/1.3066(s/\omega_1) + 1}$$

$$H_1(s) = \frac{(s/\omega_1)^2}{(s/\omega_1)^2 + 1/0.5412(s/\omega_1) + 1}$$

Let

$$\omega_1 = \frac{\omega_C Q_{TC}}{1.3066}$$

$$k = 1 - (\omega_1/\omega_C)^2$$

Equalizer transfer-function can be written as

$$F(s) = \frac{1}{1-k}(1 - kH_1(s))H_2(s)$$

$$\text{Gain} = \frac{\sqrt{k^2 + (0.7654)^2}}{\sqrt{2}(1-k)}$$

For $Q_{TC} = 0.707$, we obtain $k = 0.707$, so the amplitude should be increased into 2.52 times, or 8.01dB.

Compressor

The compressor module is introduced to limit the dynamic range of the input signal. On one hand, limiting the maximal value of the signal can protect the free membrane of the emitter from defecting, which is commonly due to excessive displacement. On the other, adjusting the input signal to a narrow range of amplitude can serve as a major countermeasure to distortion, generally improving the audio quality.

Common implementations of a compressor for a sound source take two different approaches: feedback design and feedforward design. Both are adaptive based on DSP techniques but have different advantages and disadvantages.
A feedback design comprises of a dynamic range compressor (DRC) with a non-linear gain function and a feedback control loop. Dynamic characteristics (such as temperature, current and voltage) of the signal at the output port are measured for estimation of audio quality and membrane displacement. Based on these measurements, the controller adjusts the operating parameter of the DRC so that it adapts to the current audio signal. For example, when the displacement of the free membrane is above a certain level, the compression ratio and downward threshold of the DRC may be decreased in order to lower the amplitude of the audio signal. One implementation of the feedback design is shown in (Gautama, 2012.).

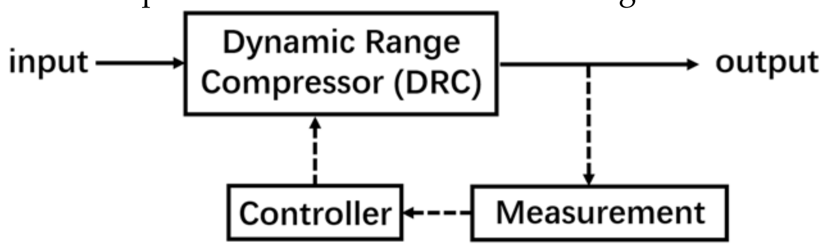

Figure 6: A feedback design of the compressor

A feedforward design comprises of a DRC and a feedforward control loop. This loop includes a similar or identical DRC and therefore can collect information of the output signal and directly send it to the controller. The controller then adjusts the operational parameters of both DRCs. One implementation of the feedforward design is shown in (Gautama, 2013.).

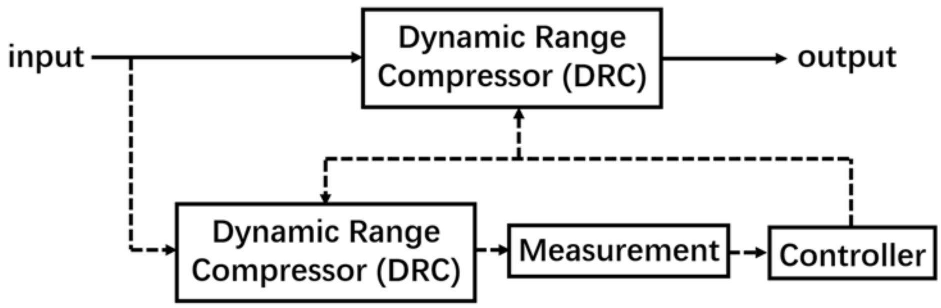

Figure 7: A feedforward design of the compressor

Comparing the two implementations above, we note that generally the feedforward approach can outperform the feedback one, since its parameter adjustment is instantaneous as opposed to having a small delay due to the feedback loop. However, the feedforward approach requires duplicate implementation of DRCs, which leads to more space and higher power consumption rate. To conclude, the actual implementation of the compressor module depends on the priority of the two factors.

LPF

The basic operation of an active LPF is the same as for its equivalent RC passive high pass filter circuit, except that the circuit has an operational amplifier, providing amplification and gain control.

In our signal processing system, LPF is used to provide a cutoff frequency, f_{cut}, of the higher frequency component, which is out of human hearing range. Besides, it can also eliminate the secondary wave, which is generated with high levels of harmonic distortion(Kite et al., 1998). The carrier frequency f_c is about 44KHz, as we will discuss later. After modulation, the range of frequency is $[f_c - f_{cut}, f_c + f_{cut}]$. As the human hearing range is from 200Hz to 20KHz, it is possible that $f_c - f_{cut}$ still falls in that, resulting in an annoying audible beat signal.

Specifically, we set the cutoff frequency f_{cut} to 15KHz~20KHz, which could eliminate the audible beat signal after modulation. We consider an active low-pass filter design in our implementation, which enjoys better performance and predictability than traditional filter[2] and avoiding use inductors (typically expensive compared to other components). The design of a common active low-pass filter is shown as FIG. 8.

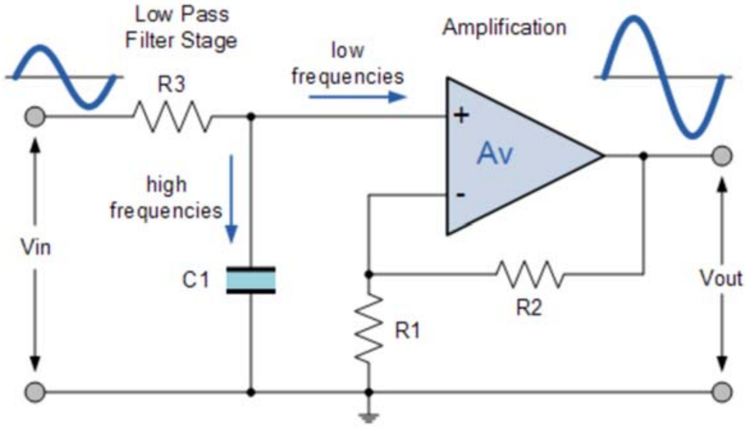

Figure 8: Active Low Pass Filter with Amplification

The shape of the response, the quality factor Q, and the tuned frequency can often be tuned with inexpensive variable resistors(Lancaster et al., 1975).

As an example, a 10th order Butterworth filter, whose stopband is 23KHz and passband is 18KHz, is shown in FIG. 9.

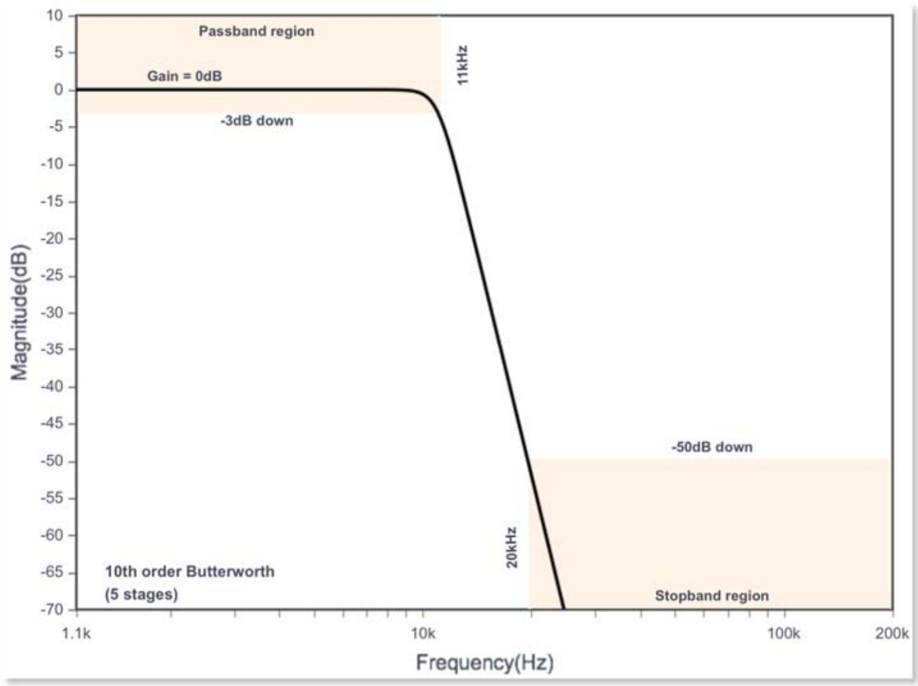

Figure 9: Frequency response of a LPF

HPF

Like the previous active low pass filter circuit, the simplest form of an active high pass filter is shown in FIG. 10.

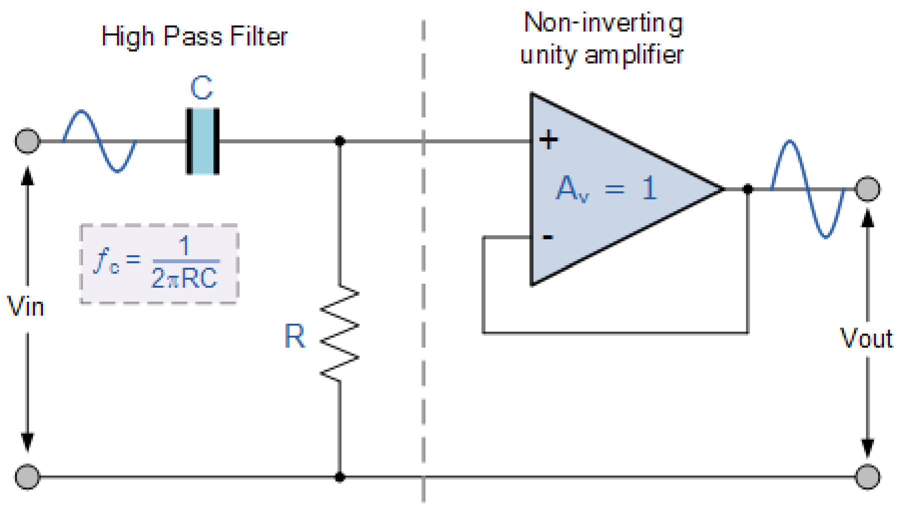

Figure 10: [First order High Pass Filter]

The high pass filter is configured to eliminate low frequencies lower than about 20-200Hz, which is out of human hearing range. Besides. It also aims to clear the frequencies that would result in deviation of carrier frequency after modulation. The frequency response of a 3th Butterworth filter, whose passband frequency is 200Hz and stopband frequency is 20Hz, is shown in FIG. 11.

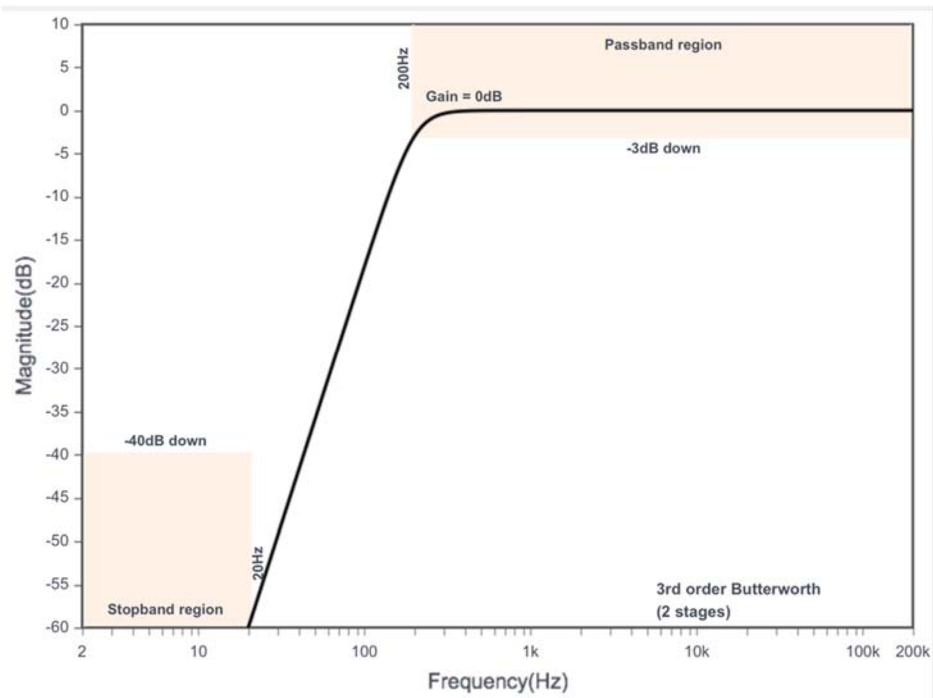

Figure 11: Frequency response of a HPF

The high pass filter, set behind modulation, is configured to pass modulated ultrasonic carrier signal and ensure that no audio frequencies enter the amplifier. The cutoff frequency is 25kHz approximately.

5. Timetable

Provide detailed information on the expected timetable for the project. Break the project into phases, and provide a schedule for each phase.

	Description of Work	Start and End Dates
Phase One	Personnel recruitment, Needs assessment, field study, materials preparation.	Apr. 1st to Apr.30th
Phase Two	(i)System Design, Prototype Building, (ii)Testing and enhancement.	(i) May. 1st to Jun. 30th (ii) Jul. 1st to Aug. 31st
Phase Three	Feedback, Productization (including miniaturization), Product Training.	Sep. 1st to Oct. 15th

Table 1: Timetable

6. Budget

State the proposed costs and budget of the project. Also include information on how you intend to manage the budget.

	Description of Work	Start and End Dates
Phase One	Labor costs: $ 13000 Materials preparation: $ 15000	Apr. 1st to Apr.30th
Phase Two	(i) Labor costs: $ 26000 Supplies: $ 1000 (ii) Labor costs: $ 26000 Additional material: $ 5000	(i) May. 1st to Jun. 30th (ii) Jul. 1st to Aug. 31st
Phase Three	Labor costs: $ 26000 Productization cost: $ 20000	Sep. 1st to Oct. 15th
	Total	**$ 132,000.00**

Table 2: Budget

7. Key Personnel

List the key personnel who will be responsible for completion of the project, as well as other personnel involved in the project.

Client	Apple Inc
Sponsor	UESTC
Project manager	Limei Xu
Team	Ziyi Yuan, Zexi Huang, Junyu Tao, Xinyu Gong, Boyu Ning.

Table 3: Key personnel

8. Evaluation

In order to complete the 5 objectives (High audio quality. Miniaturization. Low power consumption. Harsh condition resistance. Robust. Economic). We developed 5 indicators to measure it, include the system must be imbed into a smartphone like iphone7, the power consumption must under 3w, the audio quality distortion less than 1%, the system function appropriately at a temperature range of 0~40℃ and life-span must more than 4 years. In addition, we divide the phase two into 5 part, each part have 12 days to complete 20% in total project. At the end of each part, the sponsor will conduct monitoring and evaluation activities to determine project's progress and at the end of the project get the outcome. If we cannot complete the project within the prescribed time. The sponsor have right to withdraw funds.

Objectives	**Indicators**
High audio quality	Distortion less than 1%
Miniaturization	Imbed into a smartphone like iphone
Low power consumption	Power under 3w
Harsh condition resistance	Temperature range of 0~40℃
Robustness	More than 4 years

Table 4: Objects and indicators

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