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COMPARISON OF WIDEBAND EARPIECE INTEGRATIONS IN MOBILE PHONE

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The speech in telecommunication networks has been traditionally narrowband ranging from 300 Hz to 3400 Hz. It can be expected that wideband speech call services will increase their foothold in the markets during the coming years.

In this thesis speech coding basics with adaptive multirate wideband (AMR-WB) are introduced. The wideband codec widens the speech band to new range from 50 Hz to 7000 Hz using 16 kHz sampling frequency. In practice the wider band means improvements to speech intelligibility and makes it more natural and comfortable to listen to.

The main focus of this thesis work is to compare two different wideband earpiece integrations. The question is how much the end-user will benefit from using a larger earpiece in a mobile phone? To find out speaker performance, objective measurements in free field were done for the earpiece modules. Measurements were performed also for the phone on head and torso simulator (HATS) by wiring the earpieces directly to a power amplifier and with over the air on GSM and WCDMA networks. The results of objective measurements showed differences between the earpiece integrations especially on low frequencies in frequency response and distortion.

Finally the subjective listening test is done for comparison to see if the end-user notices the difference between smaller and larger earpiece integrations using narrowband and wideband speech samples. Based on these subjective test results it can be said that the user can differentiate between two different integrations and that a male speaker benefits more from a larger earpiece than a female speaker.

Keywords: AMR-WB, wideband speech, earpiece, subjective testing, acoustic measurements

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Perinteisesti puhelinverkoissa välitettävä puhe on ollut kapeakaistaista, kaistan ollessa 300 - 3400 Hz. Voidaan kuitenkin olettaa, että laajakaistaiset puhepalvelut tulevat saamaan markkinoilla enemmän jalansijaa tulevina vuosina.

Tässä lopputyössä esitellään puheenkoodauksen perusteet laajakaistaisen adaptiivisen moninopeuspuhekoodekin (AMR-WB) kanssa. Laajakaistainen puhekoodekki laajentaa puhekaistan 50-7000 Hz käyttäen 16 kHz näytetaajuutta. Käytännössä laajempi kaista tarkoittaa parannuksia puheen ymmärrettävyyteen ja tekee siitä luonnollisemman ja mukavamman kuuloista.

Tämän lopputyön päätavoite on vertailla kahden eri laajakaistaisen matkapuhelinkuulokkeen integrointia. Kysymys kuuluu, kuinka paljon käyttäjä hyötyy isommasta kuulokkeesta matkapuhelimessa? Kuulokkeiden suorituskyvyn selvittämiseksi niille tehtiin objektiivisia mittauksia vapaakentässä. Mittauksia tehtiin myös puhelimelle pää- ja torsosimulaattorissa (HATS) johdottamalla kuuloke suoraan vahvistimelle, sekä lisäksi puhelun ollessa aktiivisena GSM ja WCDMA verkoissa. Objektiiviset mittaukset osoittivat kahden eri integroinnin väliset erot kuulokkeiden taajuusvasteessa ja särössä erityisesti matalilla taajuuksilla.

Lopuksi tehtiin kuuntelukoe tarkoituksena selvittää erottaako loppukäyttäjä pienemmän ja isomman kuulokkeen välistä eroa käyttäen kapeakaistaisia ja laajakaistaisia puhelinääninäytteitä. Kuuntelukokeen tuloksien pohjalta voidaan sanoa, että käyttäjä erottaa kahden eri integroinnin erot ja miespuhuja hyötyy naispuhujaa enemmän isommasta kuulokkeesta laajakaistaisella puhekoodekilla.

Hakusanat: AMR-WB, laajakaistainen puhe, kuuloke, subjektiivinen testaus, akustiset mittaukset

Preface

This thesis has been made for Nokia Corporation Smart phones Electro Mechanics Audio Oulu unit during November 2008 – December 2009. I'd like to thank all my colleagues at Nokia, especially my instructor Panu Nevala who guided my through this whole process. I would also like to thank my supervisor, professor Paavo Alku.

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Thank you!

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Samppa Hyrkäs

Contents

Abbreviations

3G	International Mobile Telecommunications-2000 (IMT-2000)
3GPP	3rd Generation Partnership Project
ACELP	Algebraic Code-Excited Linear Prediction
AGC	Adaptive Gain Control
AMR	Adaptive Multi-Rate speech codec
AMR-NB	Adaptive Multi-Rate NarrowBand speech codec
AMR-WB	Adaptive Multi-Rate WideBand speech codec
A/D	Analog-to-Digital (conversion)
B&K	Brüel & Kjaer
CI95%	95% Confidence Interval
CELP	Code-Excited Linear Prediction
CMOS	Comparative Mean Opinion Score
dB	deciBel
D/A	Digital-to-Analog
DCT	Discrete Cosine Transform
DM	Delta Modulation
DRC	Dynamic Range Controller
DRP	Ear-Drum Reference Point
DUT	Device Under Test
EFR	Enhanced Full Rate speech codec
ERP	Ear Reference Point
ETSI	European Telecommunications Standards Institute
FIR	Finite Impulse Response
FR	Full Rate speech codec
FS	Full Scale
Glottis	The space between the vocal cords
GSM	Global systems for mobile communications
HATS	Head And Torso Simulator
IEC	International Electrotechnical Commission
IHF	Integrated Hands-Free
ISP	Immittance Spectral Pair
LP	Linear Predicting
LPC	Linear Predictive Coding
LTP	Long Term Prediction
MDRC	Multiband Dynamic Range Controller
MIPS	Millions of Instructions Per Second
MOS	Mean Opinion Score
PCM	Pulse Code Modulation
RF	Radio Frequency
RPE-LTP	Regular Pulse Excitation with Long Term Prediction
SNR	Signal-to-Noise Ratio
THD	Total Harmonic Distortion
THD+N	Total Harmonic Distortion and Noise
Vocoder	VOice CODER
VSELP	Vector Sum Excited Linear Prediction
WCDMA	Wideband Code Division Multiple Ac

1. INTRODUCTION

In telecommunication speech transmission is based nowadays on digital transmission. When speaking into the phone's microphone the human speech is in analogical form. The microphone transforms the air pressure variations to a digital form with an analog-to-digital (AD) converter and after that the speech is digital sampled data. The traditional telephone system uses a pulse code modulation (PCM) method to do this AD conversion. The bit rate of PCM requires bandwidth too much for cellular radio transmission and, therefore, the speech information has to be compressed. This compression is also called speech coding.

The aim of speech coding in transmission systems is to optimize the speech quality in relation to consumed bits and error robustness. There are many different speech coding methods. The best coding methods compress the data so that the human ear does not sense much difference to the uncompressed version of the speech. From the speech codecs standardized for the cellular telephony, adaptive multi-rate wideband speech codec (AMR-WB) produces the most natural sound.

The traditional land line and the global systems for mobile communications (GSM) network have used the speech bandwidth 300Hz-3400Hz. The first commercial network with speech coding method AMR-WB on the bandwidth 50-7000Hz was opened in Moldova at September 2009 [24]. The AMR-WB is widely expected to become a new standard for mobile voice communications. It can be expected that operators will introduce the AMR-WB voice service in many networks around the world in the near future.

This thesis focuses on comparing two different earpiece integrations in wideband and narrowband speech calls. The idea is to find out how much a user benefits if a larger earpiece is used in AMR-WB calls and in AMR-NB calls for comparison.

This paper contains information about hearing and speech, current speech coding methods on the market with the upcoming AMR-WB codec. A wideband audio path from the antenna to the loudspeaker is introduced. The loudspeaker enclosures affect on the overall acoustics and different cavities with leaks is presented.

The objective measurements are done to find out if there is any significant difference between the earpieces in free field without the phone, on the head and torso simulator (HATS) by using a connection directly from the measurement equipment to the speaker on the phone. Also, the phones' audio performances are measured during the call to evaluate the performance under the whole audio path. The subjective test is done in Oulu Nokia Teknologiakylä site to get information how users sense the difference between the two phones. Finally the objective results are compared with subjective test results, which heavily support the decision taken.

2. BACKGROUND THEORY OF SPEECH AND HEARING

This chapter introduces the speech properties and coding used in mobile communication systems.

2.1 Speech properties

Speech is an excellent way to communicate with other people. Visual effects can be used to make the speech more effective, but in telephone calls the voice is the only available way to communicate.

Humans have unique characteristics compared to other living creatures on earth: speech. It is in everyday usage and self-evident for us, but if something goes wrong then we notice how much it means to us.

The speech is acoustic sound waves from the speaker's vocal organs to the listener's ears. The smallest posited structural unit of the speech is a phoneme [1]. For example, in the Finnish language there are 24 phonemes. The understandable words are made from phonemes after another and sentences are formed by placing pauses between the sequential phonemes.

Phonemes can be divided into two groups: voiced and unvoiced phonemes. In the Finnish language, the vocal phonemes and some of the consonants (e.g. n, m, j and v) are voiced. The parts of the consonants are unvoiced (e.g. k, p, t, f, s and h). The unvoiced phonemes are noisy without periodicity. The voiced parts are periodic in the time-domain and harmonic structure in the frequency-domain. These properties are from vocal tract resonances. There are peaks in the voiced phoneme spectrum and those are vocal tract resonance frequencies called formants. The different phonemes can be distinguished by looking at the formant structure. Finnish vocal phonemes can be differentiated with the first two lowest formants. It is, however desirable that there are higher formants included when transmitting a speech signal.

2.1.1 Speech production

The speech is produced from a filtering operation, where the stimulus goes through about a 17 cm long sound channel (see Figure 1) formed by larynx, pharynx, oral and nasal cavity [2]. The sound channel is a physiological filter, which shapes the stimulus from the lungs. Different sounds are formed by changing the filtering characteristics. These properties change when the profile of the sound channel is shaped with the different position of, for example tongue and lips.

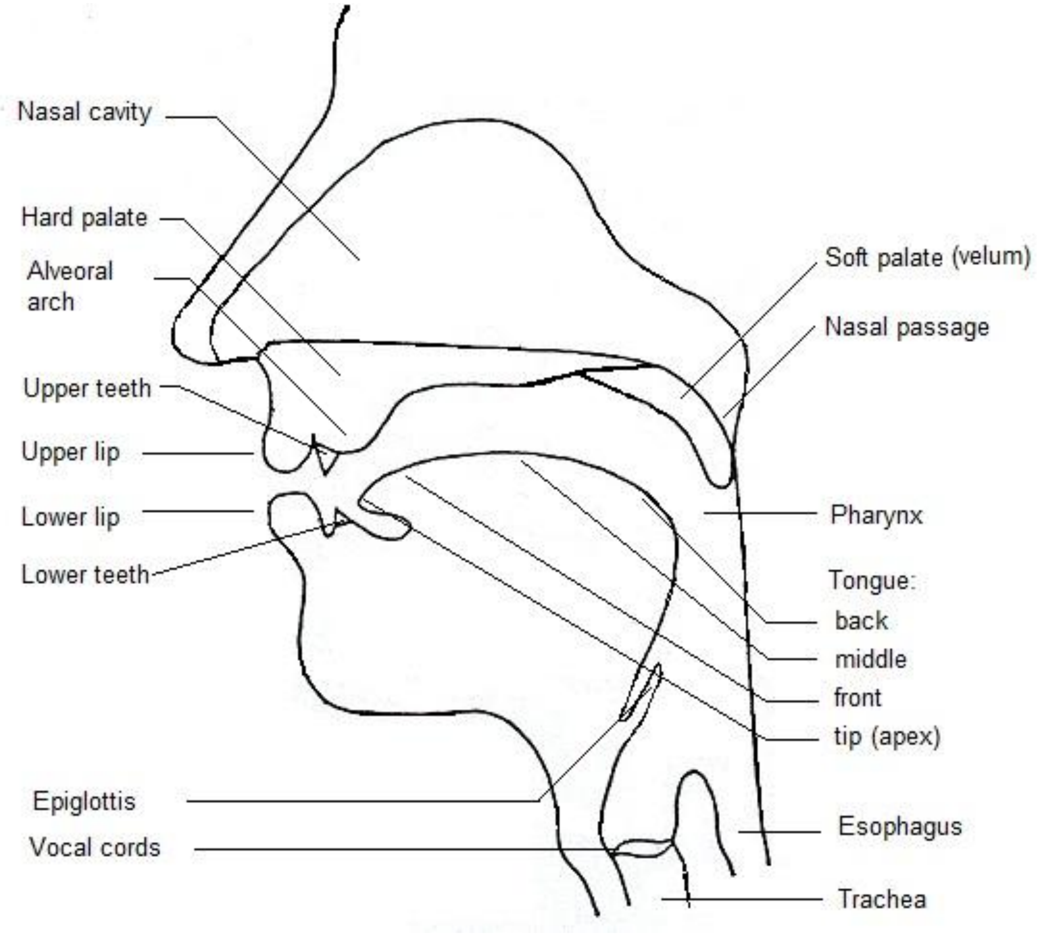


Figure 1: The human vocal organs [3].

Voiced sound forming starts from the lungs. Midriff muscle press' the lungs and causes overpressure to the trached. The vocal cords start to vibrate because the air flows from the lungs through a small hole between the vocal cords called the glottis. The vibrating frequency is called the fundamental frequency, which is about 100 – 110 Hz for males and 200 Hz for females. The periodic airflow pulse from the vocal cords is called the glottis stimulus. In the end of the sound channel, the filtered glottis stimulus diverges from the mouth and changes to audible pressure wave.

One difference between the voiced and unvoiced sound is that voiced sounds have greater amplitude than unvoiced. The waveforms of voiced sounds are exact periodic, which is very important from the speech coding point of view. Instead of using the glottis stimulus in the unvoiced sounds, they are formed in narrow or closed parts of the sound channel. Unvoiced sounds are often similar to random noise and the glottis stimulus is not used at all.

One way to present the speech production is to use a simplified source-filter model of speech as in Figure 2. This kind of model can be also used in the formant synthesis to produce synthetic speech. Voiced sounds are produced from the glottis stimulus and unvoiced from noise. Both voiced and unvoiced stimulus are connected to a binary switch. After the switch there is an input for a linear filter, which represents other parts of the speech production, especially the sound channel [4]. Gain A_0 is needed for balancing the speech signal energy on every stimulus and filter combination. The F_0in Figure 2 is the fundamental frequency of voiced sounds.

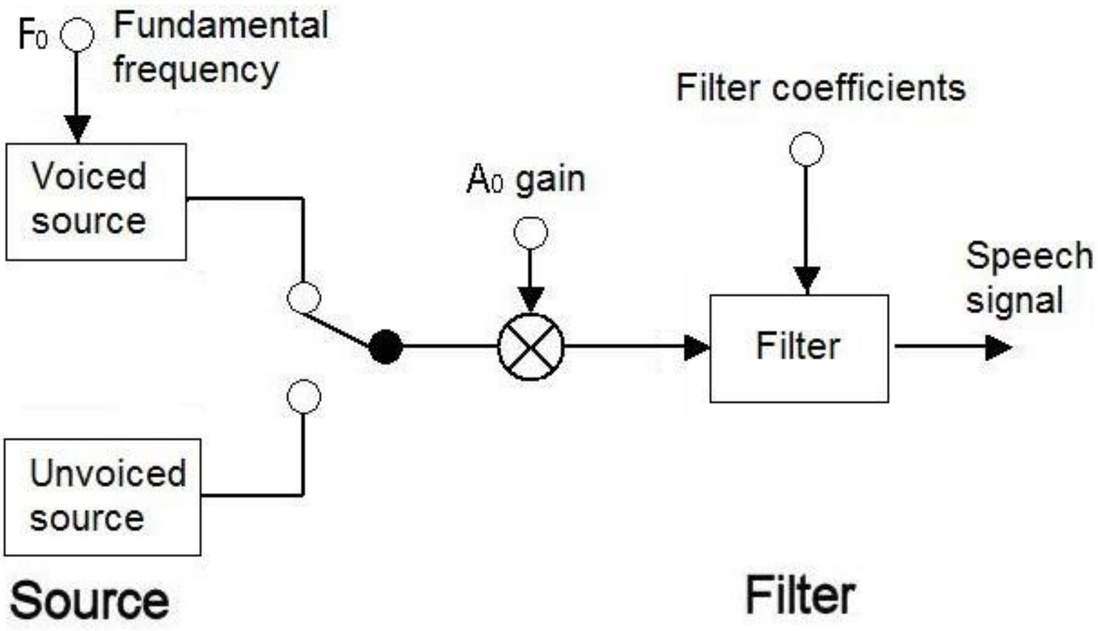


Figure 2: Source-filter model of speech [1].

2.1.2 Hearing of speech

The role of the ear is to receive the sound wave from the air and guide it to the hearing nervous system. The sensitivity of the human ear is not always so good compared to an animal ear, but it has a unique special assignment and ability: speech analysis and recognition. The structure of the human ear can be seen in Figure 3.

The human ear is usually divided into three parts: inner, middle and outer ear. The auditory canal ends to the tympanic membrane in the outer ear. Inside the eardrum in the middle ear is three bones connected to each other: The hammer, anvil and stirrup. These bones transmit and strengthen the sound, but also prevent the wide eardrum movement effect to the inner ear. The purpose of the middle ear is to adjust the impedances between the air in the outer ear and the liquid in the inner ear. The three bones mentioned above act as a mechanical impedance converter by transferring low pressure and high particle speed (in air) into high pressure and low particle speed (in liquid) [1].

The speech recognizing and understanding starts in the inner ear. The cochlea is a very sensitive organ, which analyzes the sound and transforms the sound to nerve impulses. The semicircular canals are not for hearing, but for human balance.

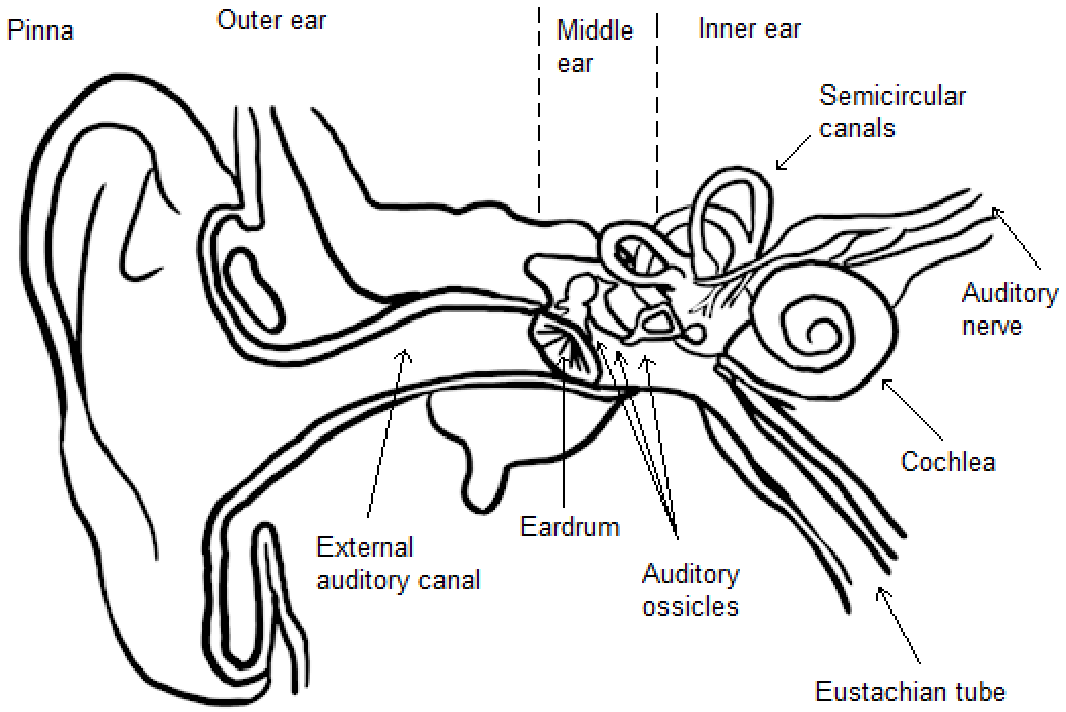


Figure 3: The structure of human ear [7].

3. FUNDAMENTALS OF GSM CODERS

3.1 Speech coding basics

The idea of speech coding is to compress the original sound data as much as possible without losing the quality too much [4]. This compression enables the speech transmission with smaller resources and increases the amount of information to be transmitted with limited resources.

When comparing audio formats, the sound sources have to be selected carefully. The normal audio CD has a very good audio quality and to the present day it has been the standard physical medium for sale of commercial audio recordings. If the CD audio is transmitted with stereo sound, it takes over 1.4 Mbits/s transfer speed. The bandwidth in CD is half from the sampling rate 44.1 kHz/2 = 22 kHz. In a traditional telephone system, the used bit rate is 64 kbits/s, which is about 1/22 of CD audio data rate called PCM. Usually the CD audio quality is compared to all audio formats, where the PCM is used to grade speech audio quality.

Before the analog speech is coded to PCM format some signal processing must be done. First the original speech signal is filtered and the unwanted signal components are removed. The traditional telephone network uses a bandwidth between 300 Hz – 3400 Hz so the frequencies outside the band are filtered away. After filtering the speech signal is then sampled. This means taking samples of the signal at the sampling frequency, which is in the PCM case 8 kHz. When sampling is done, the signal values are transformed to discrete numerical values or quantized. The quantization is done with 13 or 14 bits in telephone networks. Finally, after the quantization the signal is coded. Signal coding reduces the bits needed for data transmission for instance compressing with A- or µ-law decreases the sample to 8 bit with a very small signal quality loss. The basic principle of speech coding is shown in Figure 4.

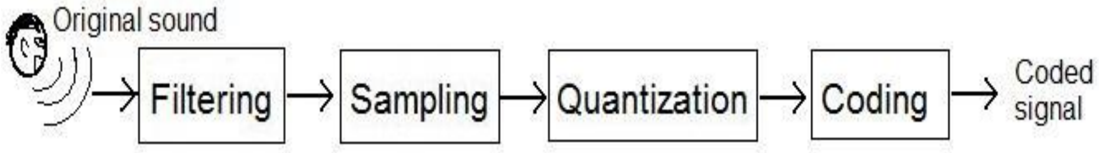


Figure 4: Speech coding principle from original sound to coded signal.

Speech coding methods are divided into two groups: waveform coders and source coders. The waveform coders try to transmit the original signal to the destination and keep the same waveform. The idea of source coders is to model the mechanism how the waveform is produced by parameters. There are also hybrid coding methods that connect both of these methods.

3.1.1 Waveform coders

Most of the sounds that we hear are vibrations through air. These vibrations can be transformed into electrical signals with the help of the microphone. The microphone signal is then coded on the desired media. Waveform coders try to preserve the electrical signal waveform as much as possible.

The advantage of the waveform coders is that they can be applied to different kinds of signals like music, signaling or data transfer. If there is noise added to the signal, waveform coders maintain their performance.

The simplest waveform coding method is pulse-code modulation PCM. It is standardized in ITU-T G.711 [5]. PCM transforms the linear 13 or 14 bit samples to 8 bit one by one according to the standard. Using PCM guarantees very good speech quality, but bit rate is fairly high.

There are several ways to improve pulse-code modulation performance. One is to use differential modulation, which is based on prediction of the samples of the signal and baseline of PCM. Another method is adaptive quantization, where the size of the quantization step is varied allowing the reduction of the required bandwidth for a given signal-to-noise ratio. These two coding methods can achieve almost as good speech quality as PCM, but with a smaller bit rate.

There are also other waveform coding methods for example, delta modulation and adaptive transform coding. The latter method uses fast transforming algorithms, like discrete cosine transform DCT to cut the signal on a large amount of frequency bands. The bit amount of frequency band multipliers is selected based on the speech spectrum. Delta modulation is one variant of PCM, which uses a very low bit amount to indicate the change of the previous sample.

3.1.2 Source coders

Vocoders or source coders are developed to achieve efficient speech coding. The speech signal is sent to the transmission channel as parameters reducing the bit rate noticeably. Even the speech is in parameter form, it can be reconstructed in the receivers end so that the human ear senses characteristic parts of the original signal. Most of the vocorders are based on the speech production model in Figure 2.

There are many flaws in source coders. Because the vocoders are optimized to speech coding, other types of signals suffer more in the coding. Usually the speech quality is worse and more synthetic than waveform coders. Vocoders are also talker dependent and male voices are typically heard with better quality. If there is noise added to the signal, the quality of the coded speech decreases recognizably. Most of the speech coders used in telecommunication are based on linear prediction and its variations.

3.1.3 Present speech codecs

There are many speech codecs available for speech compressing. The most common codecs are listed in *Table 1*. There are several codecs used in mobile communication for example typically, the supported codecs in mobile phones are GSM HR, GSM FR, GSM EFR and GSM AMR. Also AMR-WB is specified for GSM and WCDMA and the first commercial AMR-WB network was launched to consumers on autumn 2009 [24].

In *Table 1* the Mean Opinion Score, MOS means the average quality that listeners perceive in a listening test on a scale of 1-5. Values from 4.0 to 4.5 are as good as telephone land line, mobile networks are graded 3.5-4.0 and values 2.5-3.5 sound like synthetic speech [8].

Table 1: Most common speech codecs [6]

Codec	Coding method	Bit rate (kbit/s)	MOS	Complexity MIPS
AMR WB (G.722.2)	ACELP	6.60 - 23.85	WB	40
G.722	SB-ADPCM	48 / 56 / 64	WB	5
G.711	PCM	64	4.4	0.5
G.726	ADPCM	16 / 24 / 32 40	2 / 3.2 / 4.0 / 4.2	2
G.727	E-ADPCM	16 / 24 / 32 40	2 / 3.2 / 4.0 / 4.2	2
AMR	ACELP	4.75 - 12.2	≤ 4.2	17
GSM EFR	ACELP	12.2	4.2	16
CDG27	QCELP13	1.0 / 6.2 / 13.3	4.1	
IS-127	ACELP (EVCR)	0.8 / 4 / 8.55	4.1	24
G.728	LD-CELP	16	4	30
IS-641	ACELP	7.4	4	15
G.723.1	A/MP-MLQ CELP	5.2 / 6.2	3.7 / 4.0	16
G.729	CS-ACELP	8	3.9	20
G.729a	CS-ACELP	8	3.7	11
GSM FR	RPE-LTP	13	3.7	5 - 6
GSM HR	VSELP	5.6	3.6	14
IS-54	VSELP	7.95	3.5	14
IS-96-B	QCELP	0.8 / 2 / 4 / 8.55	3.5	15
Inmarsat-Aero	MPLPC	8.9	3.5	
TETRA	ACELP	4.56	< 3.5	15
JDC	VSELP	6.7	< 3.5	
Inmarsat-M	IMBE	4.15	< 3.5	7
Inmarsat-P	AMBE	3.6	< 3.5	
DOD FS 1016	CELP	4.8	3.2	16
DOD FS prop.	MELP	2.4	3.2	40
Inmarsat-B	APC	9.6 / 12.8	3.1 / 3.4	10
JDC-HR	PSI-CELP	3.45	< 3.0	
DOD FS 1015	LPC-10	2.4	2.3	7

The complexity in *Table 1* describes how many million instructions per second MIPS are calculated. This parameter tells how much the processor requires calculation and that way causes computational delay. The processing delay is always minimized during the designing process.

One impact, which is not mentioned in *Table 1*, is memory consumption. It affects the complexity, but it is not noted in this case. The other delays, which are not in *Table 1*, are algorithm, multiplexing and transmission delay. These delays are about the same for each speech codec used in mobile networks and for that reason are left out from *Table 1*.

3.2 Linear prediction in speech coding

Linear predictive coding (LPC) is one of the most powerful speech analysis techniques, and one of the most useful methods for encoding good quality speech at a low bit rate. It provides extremely accurate estimates of speech parameters, and is relatively efficient for computation. Almost all present speech codecs are based on this method [4].

3.2.1 Basics of linear prediction

LPC starts with the assumption that the speech signal is produced by a buzzer at the end of a tube. The glottis produces the buzz, which is characterized by its intensity (loudness) and frequency (pitch). The vocal tract (the throat and mouth) forms the tube, which is characterized by its resonances, which are called formants.

LPC analyzes the speech signal by estimating the formants, removing their effects from the speech signal, and estimating the intensity and frequency of the remaining buzz. The process of removing the formants is called inverse filtering, and the remaining signal is called the residue.

The numbers which describe the formants and the residue can be stored or transmitted somewhere else. LPC synthesizes the speech signal by reversing the process: use the residue to create a source signal, use the formants to create a filter (which represents the tube), and run the source through the filter, resulting in speech.

3.2.2 Short term prediction

In LPC analysis the sequentially placed samples' correlation is utilized efficiently. The signal sample value is estimated by forming a linear combination of a few previous samples. In linear combination, these previous samples are multiplied by certain parameters. When the multipliers and products are added up, the prediction to the sample value is obtained. This value is subtracted from the sample value and the prediction error, and the residue is attained as a result.

The prediction is repeated to a certain amount of sequential samples using the same coefficient parameters. After this, the square error between the original and predicted signal samples is minimized. The result shows the optimal coefficient parameters. The filter is a finite impulse response (FIR) type and called the prediction filter.

3.2.3 Long term prediction

The long term prediction filter estimates the coming residual peaks at the end of the pitch-period and removes the peaks with inverse filtering. After inverse filtering the new residual is more like hum, which can be quantized with a small amount of bits.

The short term predictor's prediction error signal is like an impulse, which is from the voiced speech signal glottis pulses. Describing the impulse signal by a low amount of data bits is problematic, which is why the long term predictor is added after short term prediction filter.

3.2.4 Optimization of prediction filter

In general form the LPC is done at p-degrees, when samples *s(n)* prediction $\check{s}(n)$ calculation is done with *p* previous samples (*s(n-1), s(n-2),…,s(n-p)*). The coefficient parameters are marked on *a(k)*. The expression for prediction is obtained in (1).

$$\check{s}(n) = \sum_{k=1}^{p} a(k)s(n-k) \qquad (1)$$

The prediction error called residual can be expressed as:

$$e(n) = s(n) - \check{s}(n) = s(n) - \sum_{k=1}^{p} a(k)s(n-k) \qquad (2)$$

In the infinite length time window, the residual signal energy is expressed as:

$$E = \sum_n e^2(n) = \sum_n \left[s(n) - \sum_{k=1}^{p} a(k)s(n-k)\right]^2 \qquad (3)$$

$$= \sum_n [\, s^2(n) - 2s(n) \sum_{k=1}^{p} a(k)s(n-k)$$

$$+ \left[\sum_{k=1}^{p} a(k)s(n-k)\right]^2] \qquad (4)$$

The coefficients *a(k),* $1 \le k \le p$ that realize the mean square error criterion is attained when the residual energy's partial derivatives are set to zero with regard to *a(i)*:

$$\frac{\partial E}{\partial a\,(i)} = 0, 1 \le i \le p \tag{5}$$

$$\sum_n \left[-2s(n)s(n-i) + 2\sum_{k=1}^{p} a(k)s(n-k)s(n-i)\right] = 0 \tag{6}$$

$$\sum_n s(n)s(n-i) = \sum_{k=1}^{p} a(k)s(n-k)s(n-i), 1 \le i \le p \tag{7}$$

The equation (7) can be expressed in the following form:

$$\sum_{k=1}^{p} a(k)\phi(i,k) = \phi(i,0),\ 1 \le i \le p, \tag{8}$$

where $\phi(i,k) = \sum_n s(n-i)s(n-k)$

The optimized prediction filter gives the residual energy:

$$E_{min} = \phi(0{,}0) - \sum_{k=1}^{p} a(k)\phi(0,k) \tag{9}$$

$$= \sum_n s^2(n) - \sum_{k=1}^{p} a(k)\left(\sum_n s(n)s(n-k)\right) \tag{10}$$

$$= E_{orig} - E(p) \tag{11}$$

Equation (11) shows that residual energy E_{min} can be expressed as the subtraction of original signal E_{orig} and prediction degree dependend energy $E(p)$ [8].

3.2.5 Windowing and autocorrelation method

There are two ways to calculate LPC, the autocorrelation and covariance methods. The autocorrelation method is most often used because it requires less calculation and the FIR-filter is always at a minimum phase after optimization. The minimum phase filter is necessary so that the infinite impulse response (IIR) filter decoder is stable.

In theory, the FIR-filter optimization is done in an infinite length time frame, but in practice calculation of the speech signal is divided into short segments. The division into segments is done by multiplying the speech signal on a window function, which is nonzero on the time frame $0 \le n \le N-1$. The simplest window function is rectangle *(w(n)=1,* $0 \le n \le N-1$), which divides the signal into N sample long segments. The most common window functions in LPC are the Hanning and Hamming windows described below.

Hamming: $$w(n) = 0.54 - 0.46\cos\left(\frac{2\pi n}{N-1}\right), 0 \le n \le N-1 \tag{12}$$

Hanning: $$w(n) = \frac{1}{2}\left[1 - \cos\left(\frac{2\pi n}{N-1}\right)\right], 0 \le n \le N-1 \tag{13}$$

If these two windowing methods (11), (12) are compared to the rectangular window the advantage is that Hamming and Hanning decrease the unwanted transitions from the beginning and the end of the signal frame. The shapes of Hamming and Hanning windows are about the same; the functions gain their maximum value in the middle of the time window and their minimum values near zero at the beginning and the end.

The windowing produces the following signal:

$$s(n) = s_0(n)w(n) \tag{14}$$

where, $s_0(n)$ is original speech signal, which is continuous and nonzero, *w(n)* is windowing function, which is nonzero at time interval $0 \leq n \leq N-1$, *s(n)* describes the speech signal predicted in LPC-analysis.

Now the equation (8) term ϕ can be shown as follows:

$$\phi(i,k) = \sum_{n=o}^{N-1+p} s(n-i)s(n-k) \tag{15}$$

where $1 \leq i \leq p$ and $1 \leq k \leq p$.

When *n-i=j* is placed to Equation (15)

$$\phi(i,k) = \sum_{j=-i}^{N-1+p-i} s(j)s(j+i-k), \tag{16}$$

where $1 \leq i \leq p$ and $1 \leq k \leq p$. *s(j)* is nonzero between $0 \leq j \leq N-1$ because of the windowing and for this reason the Equation (16) can be presented as:

$$\phi(i,k) = \sum_{j=0}^{N-1-(i-k)} s(j)s(j+i-k), \tag{17}$$

where $1 \leq i \leq p$ and $1 \leq k \leq p$.

Equation (17) is the definition of the autocorrelation:

$$\phi(i,k) = R(i-k) \tag{18}$$

$$= R(k-i) = \sum_{j=0}^{N-1-(i-k)} s(j)s(j+i-k) \tag{19}$$

The result of the optimization can be represented in matrix form:

$$\mathbf{R} \bullet \mathbf{A} = \mathbf{R'}, \tag{20}$$

where **R** is autocorrelation matrix:

$$\boldsymbol{R} = \begin{bmatrix} R(0) & R(1) \cdots & R(p-1) \\ \vdots & R(0) \ \ddots & \vdots \\ R(p-1) & R(p-2) \cdots & R(0) \end{bmatrix}$$

A is a *p x 1* size vector with optimal coefficients

$\mathbf{A} = (\mathrm{a}(1), \mathrm{a}(2) \ldots \mathrm{a}(\mathrm{p}))^{T}$

R′ is autocorrelation vector:

$\mathbf{R'} = (\mathrm{R}(1), \mathrm{R}(2) \ldots \mathrm{R}(\mathrm{p}))^{T}$

The matrix A can be solved from equation (20):

$$\mathbf{A} = \boldsymbol{R}^{-1} \bullet \mathbf{R'} \tag{21}$$

Choosing the predictors degree is quite simple. When using a speech signal the degree is chosen by dividing the sample frequency by one thousand and adding a small integer. Because mobile networks use sample frequency of 8 kHz, the value for p is 8-12. Also the frame size *(N)* and window function must be chosen. Usually the *N*-value is 100-200 [8].

3.2.6 LPC-synthesis

The LPC synthesis is the reconstruction of the signal which underwent LPC analysis. It is achieved by using the stored parameters obtained from LPC analysis. When Equation (8) is solved it gives the solution to the prediction filter:

$$A(z) = 1 - \sum_{k=1}^{p} a(k) z^{-k} \tag{22}$$

If the speech signal *s(n)* is filtered through *A(z)* it gives the same residual as in (2). The idea is to code the speech signal information to optimal solved prediction filter coefficients and a residual. The coefficients and residual are sent to the receiver end with a very low bit rate compared to the waveform type coder PCM. When the coefficients and residual are sent to the receiver end, the inverse LPC-synthesis is done. The residual is filtered in LPC-synthesis with an IIR-type filter and the result is an original speech signal. The synthesis is described the in time domain:

$$s(n) = e(n) + \sum_{k=1}^{p} a(k) s(n-k) \tag{23}$$

In the z-domain:

$$S(z) = E(z) \cdot \frac{1}{A(z)} = E(z) \cdot H(z) \tag{24}$$

The residual can be quantized with a very low bit rate, because it is like noise. This property is one of the key things why LPC is such an important method when transferring a speech signal. The LPC can be described by the model found in Figure 2. The filter system coefficients are updated for every speech frame so that the sound frequency attributes are recognizable. The coefficients are retrieved from LPC analysis. The voiced sounds fundamental frequency, selection of voiced or unvoiced frame and amplification factor *G* are sent in other parameters [8].

3.2.7 Code-Excited Linear Prediction (CELP) speech coding

The most used analysis-by-synthesis coding method by recent coders is Code excited linear prediction, CELP. The idea is quite old, because Atal and Schroeder introduced the CELP in 1984 [10]. The advantage of CELP is that it offers high quality speech at a low bit rate, but the weakness is intensive computation. The algebraic code excited linear prediction ACELP vocoder algorithm is based on the CELP coding model, but ACELP codebooks have a specific algebraic structure imposed upon them. The ACELP is used in GSM enhanced full rate speech codec (EFR) and the adaptive multi-rate (AMR) speech codec.

The difference between CELP and speech production models (see Figure 2) used by other vocorders is the excitation sequence. Instead of quantizing scalar noise stimulus, the noise stimulus is viewed as a certain length vector. The CELP coder utilizes a codebook which includes a set of speech vectors, typically 256, 512 or 1024 vectors. The vector calculated from the noise stimulus is compared to codebook vectors and the best matching one is selected. The speech signal compression is achieved by sending the index of the selected vector, its scaling factor, LPC coefficients and LTP parameters to the receiver. The simplified block diagram is shown in Figure 5.

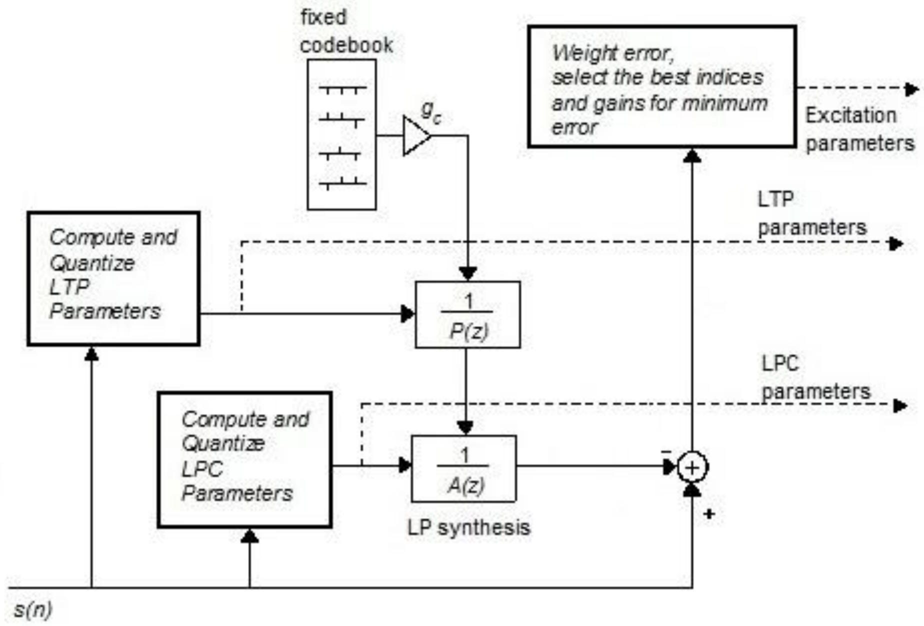

fixed codebook
g_c
Weight error, select the best indices and gains for minimum error
Excitation parameters
LTP parameters
Compute and Quantize LTP Parameters
$\frac{1}{P(z)}$
LPC parameters
Compute and Quantize LPC Parameters
$\frac{1}{A(z)}$
LP synthesis
-
+
s(n)

Figure 5: Simplified block diagram of the CELP analysis model [9]. The speech signal is marked s(n) and the fixed codebook gain factor g_c.

CELP coder has short and long-term LPC predictors. In LPC analysis the short-term prediction is done for full length speech frames by 20 ms long frames. Long-term prediction, on the other hand, is done on shorter sub frames, which are 5 ms long. The long-term prediction can be done on the original speech signal (closed-loop method) or the residual of short-term prediction (open-loop method).

Choosing the optimal excitation vector in CELP speech coding is carried out using an analysis-by-synthesis technique. First the speech is synthesised for every entry in the codebook. When the selection is done, the codeword that produces the lowest error is chosen as the excitation. There are N_f vectors in the fixed codebook and every vector has N_{SF} samples. All vectors include random sample sequence. The parameters of the LPT predictor can be held as an adaptive codebook made of N_{SF} samples. The stimulus vector can be estimated by the following equation:

$$u(n) = \hat{g}_p v(n) + \hat{g}_c c(n), \qquad (25)$$

where the gain factor of $\hat{g}_p$ is the fixed codebook and $\hat{g}_c$ is the gain factor for the adaptive codebook. The variables *v(n)* and c*(n)* are vectors from codebook.

The CELP synthesis model is presented in [14] and in Figure 6 it is realized based on the *Figure 2* speech production model. First, the adaptive codebook formed from the LPT- predictor’ parameters act as a source for predictable voiced sounds. The fixed codebook is a source for unvoiced sounds. The LPC-synthesis filter coefficients are from LPC-parameters. Finally, the post filter removes the pre-emphasis from the speech signal.

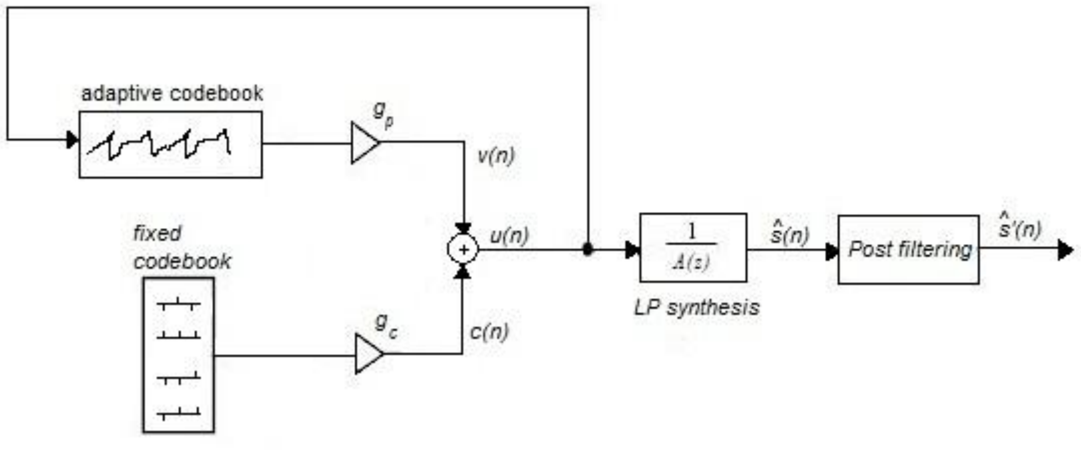


Figure 6: Simplified block diagram of the CELP synthesis model. g_p describes the gain factor of adaptive codebook, g_c is the gain factor of fixed codebook, v(n) and c(n) are codebook vectors. When c(n) and v(n) vectors are added, the result u(n) is the stimulus vector for LP synthesis. After LP synthesis $\hat{\boldsymbol{s}}(\boldsymbol{n})$ the post filtering is done and synthesis is complete $\hat{\boldsymbol{s}}'(\boldsymbol{n})$ [14].

CELP requires heavy computation and because of the codebooks high memory capacity. In order to adapt CELP for instance to mobile phones, the memory consumption and computation must be smaller and for that reason there are many variations of CELP, like algebraic code excitation linear prediction ACELP. The advantage of ACELP is that it uses the algebraic codebook, where stimulus vector search is done in a smaller vector library and codebook vectors are not saved for the sender and transmitter.

4. ADAPTIVE MULTI-RATE WIDEBAND CODEC AMR-WB

In this chapter the Adaptive Multi-Rate speech codec is introduced. The history and technical parts are described in brief.

4.1 History and standardization

The European Telecommunications Standards Institute, ETSI started a multi-rate speech codec standardization program for GSM in 1997. In 1996 the enhanced full rate, EFR codec achieved the same speech quality as in traditional landline speech and at same time was able to operate with the existing infrastructure. Even though the quality of speech was good, the need for error robust speech codec still existed. The need led to adaptive multi rate (AMR), which has an advantage that it can allocate data between speech coding and channel coding according to network conditions. The ETSI standardization program in 1997 was also a competition and winner selection was based on quality, complexity and impact on equipment and time schedule. The winner was the GSM EFR based codec developed jointly by Nokia, Ericsson and Siemens [9]. The Third Generation Partnership Project (3GPP) defined the AMR speech codec as a mandatory speech codec for third generation networks.

The AMR-WB codec standardized by ETSI/3GPP in December 2001 is jointly developed by Nokia and VoiceAge [35]. Later in January 2002 it was approved by the ITU-T as G.722.2. Before the standardization, a feasibility study and a two-phase competition were implemented to find the best codec available. Nokia implementation won the competition and beat the other competitors with a clear margin.

4.2 General description of AMR-WB

Traditional landline and GSM speech use the frequency band 300-3400 Hz providing a quality referred to as toll quality. The new speech codec AMR-WB band is more than doubled as can be seen from Figure 7. The sampling rate is increased to 16 kHz from AMR-NB 8 kHz and, therefore, the frequency range is possible to extend to the 50-7000 Hz area. Adding the lower frequencies to the speech the naturalness, presence and comfort is increased whereas high frequencies help to differentiate fricative sounds like "*f*" and "*s*". The human hearing threshold curve is also very sensitive at frequencies between 3400-7000 Hz.

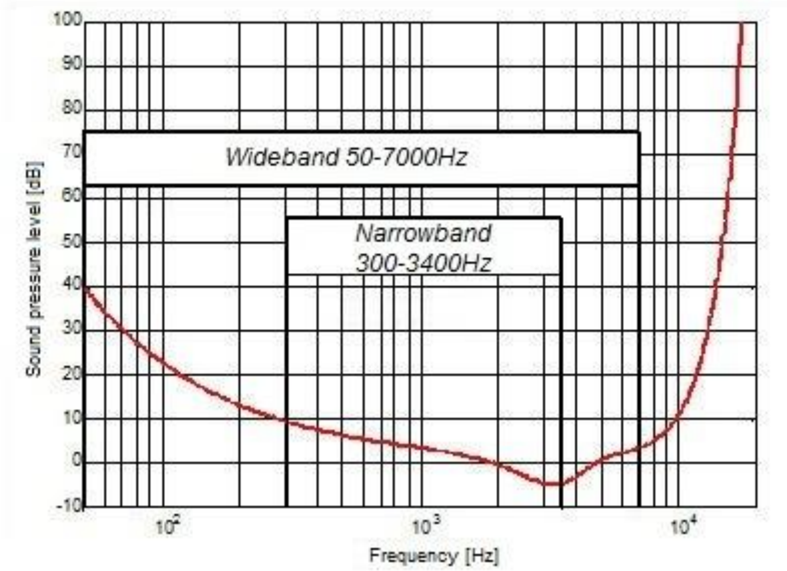


Figure 7: The hearing threshold of human auditory system with narrowband and wideband frequency range.

AMR-WB is based on the ACELP codec mentioned in section 3.2.7. The codec consists of nine source coders operating on the following bitrates 23.85, 23.05, 19.85, 18.25, 15.85, 14.25, 12.65, 8.85, 6.60 kbit/s. The 12.65 kbit/s is the lowest bit rate, which offers high quality wideband speech and the two lower bitrates are meant to be used only in temporary severe network conditions. The AMR has the ability to change the bit rate during a call and the change works differently in GSM and WCDMA networks. In GSM, the bit rate adaptation is done to provide best possible speech quality in various network conditions. If radio conditions get worse, more bits are allocated to error detection and correction, while lower codec mode is switched on improving overall quality.

Unlike in GSM, the network capacity optimization in WCDMA is done by different codec modes. However, the mode adaptation is not needed because of radio conditions because the same effect is achieved by increasing transmit power. The overall network performance and codec mode usage is monitored by the operator, which can adapt codec mode usage to maximize throughput and avoid network crowding.

One important advantage of AMR-WB is that it dynamically adapts to the traffic conditions. The variable bit rate benefit can be easily seen in Figure 8, where a 30 second call with varying channel error rate is described. The blue curve is carrier-to-interference ratio, which describes the guaranteed transmission speed. The red curve demonstrates how the AMR-WB speech codec handles the different network conditions; it changes the bit rate of the speech codec according to channel errors and this way manages to maintain network connection much better under poor conditions than constant bit rate codecs.

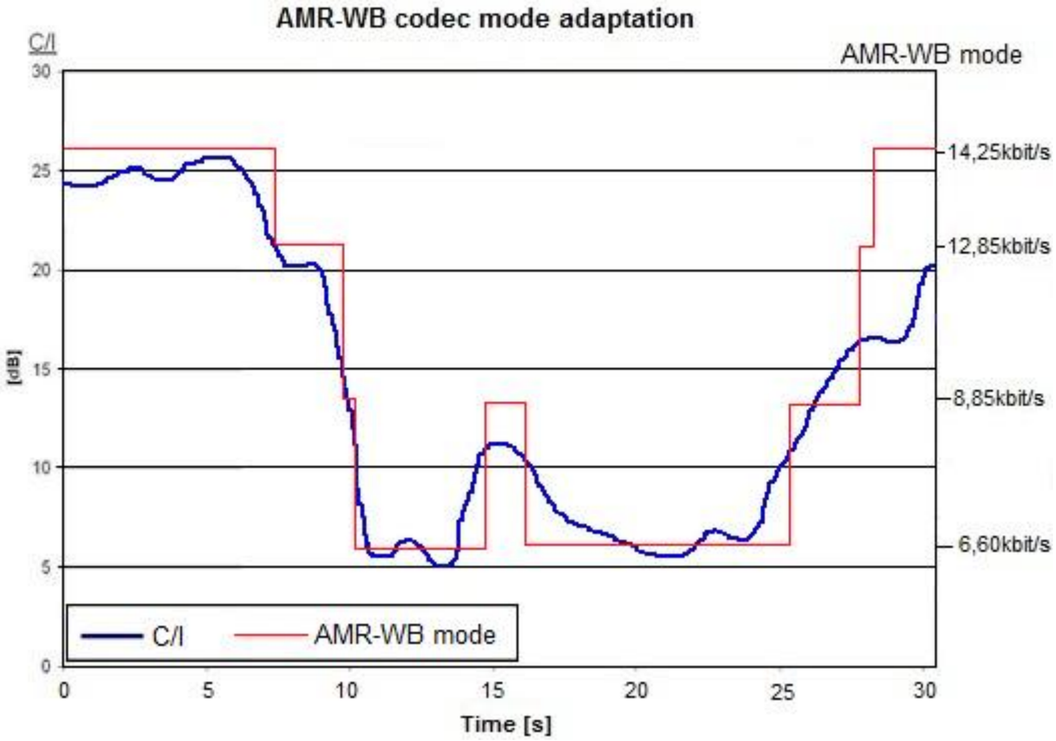


Figure 8: AMR-WB codec mode adaptation in GSM full rate channel (Carrier-to-Interference ratio, C/I) [11].

The channel error repairing is managed by power control. This means that if the radio conditions are weak, the transmission power is increased. The benefit from changing bit rate is the increased network capacity to handle more customers in peak periods by decreasing the speech coding bit rate.

The bit rate in AMR-WB can be selected asymmetric. It means that during a phone call the uplink bit rate from a phone towards the base station can be different to the downlink from the base station to phone. The speech frame length is 20 ms and the operating mode can be changed often. When the network conditions change and the suitable operating mode has to be selected, the phone uses the autonomous mode forcing the bit rate to a different level.

4.3 Encoder

The AMR-WB speech codec is based on ACELP, which means that the AMR coding method belongs to source coders. Source coders have lots of computation in their coder and the six phased block diagram of AMR can be seen in Figure 9. Pre-processing is done to all speech frames, as well as short-term LPC prediction and speech fundamental frequency analysis. The codebook searches are performed in sub frames. The specific information of the coder can be found from [14].

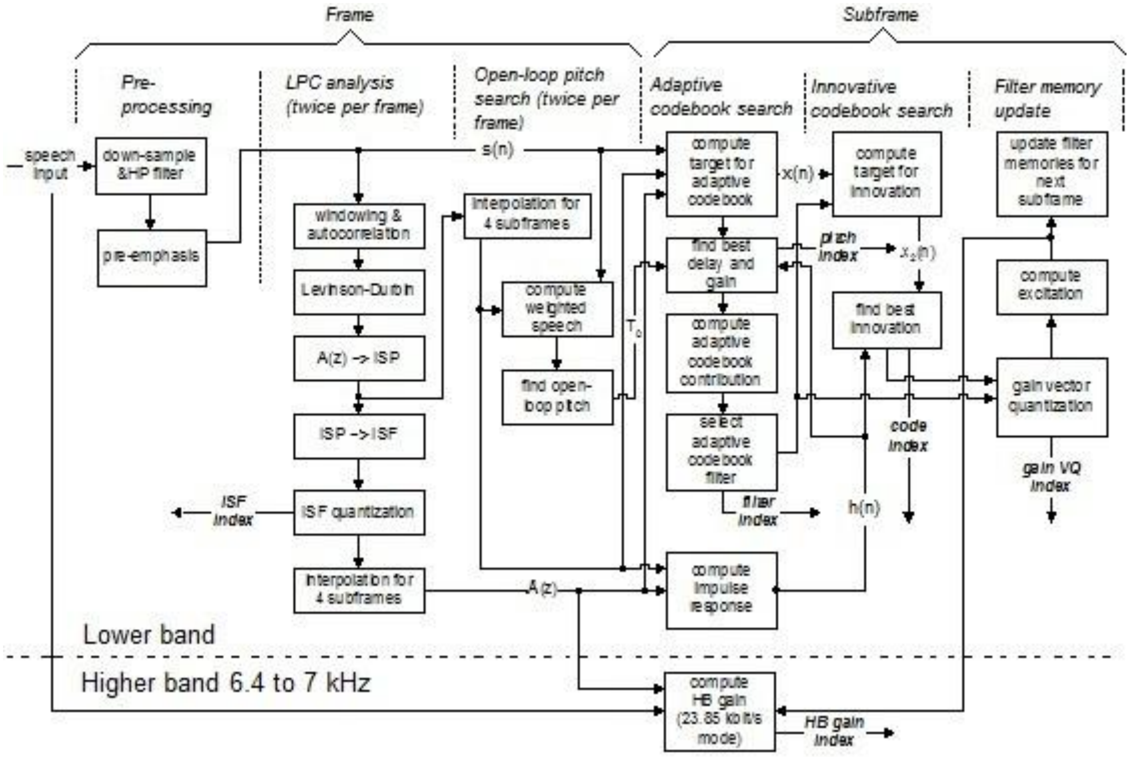


Figure 9: *Simplified block diagram of the GSM Adaptive Multi-Rate encoder [14], [32].*

In pre-processing the high pass filtering and signal level decreasing are performed. The reason for high pass filtering is that the unwanted low level signal components are removed. The analysis of the LPC, LTP and fixed codebook parameters at a 12.8 kHz sampling rate are performed. Thus a 16 kHz input signal has to be decimated.

The linear predictive analysis (LPC) is a way to approximate a speech sample as a linear combination of past speech samples. By minimizing the sum of the squared differences between the actual speech samples and the linearly predicted ones, a unique set of predictor coefficients can be determined. The short-term prediction is performed using an autocorrelation function with a 30 ms asymmetric window. To help computation, 5 ms in both directions from a window is used as an overhead.

The pitch-lag is related to the speech fundamental frequency analysis or long term prediction analysis. Accurate estimation of the pitch-lag parameter is important for the subjective quality of the synthesized speech. The search for pitch-lag parameter is divided into two parts: The approximation for pitch-lag value is found with an open-loop pitch search, which speeds up and limits the closed-loop pitch search done in an adaptive codebook search. The closed loop refines the open-loop result by finding the optimal value in the neighborhood of the open-loop result. The parameters for the adaptive codebook are the delay and gain of the pitch filter. An open-loop pitch search is done in every other sub frame in the adaptive multi-rate codec. Instead of frequency, the time between the voiced sound pulses is measured.

Synthesis and weighting filters are updated for calculating the next sub frame stimulus signal in a filter memory update. The adaptive and fixed codebook gains are vector quantized using a 6 or 7 bit codebook.

Finally, the speech frame is complete and the result is an amount of parameters quantized with a certain accuracy. The parameters are used in the receiver end in the decoder and a signal similar to the original is reconstructed.

4.4 Decoder

The function of the decoder is to decode the transmitted parameters and try to obtain the reconstructed speech by performing the synthesis. After the decoding the reconstructed speech is post filtered and upsampled. Finally, the high-band signal from 6.4 to 7 kHz is generated and added to the lower band signal. The AMR-WB decoder is based on the ACELP synthesis model and is simpler than an encoder, meaning less computation. The simplified block diagram is shown in Figure 10.

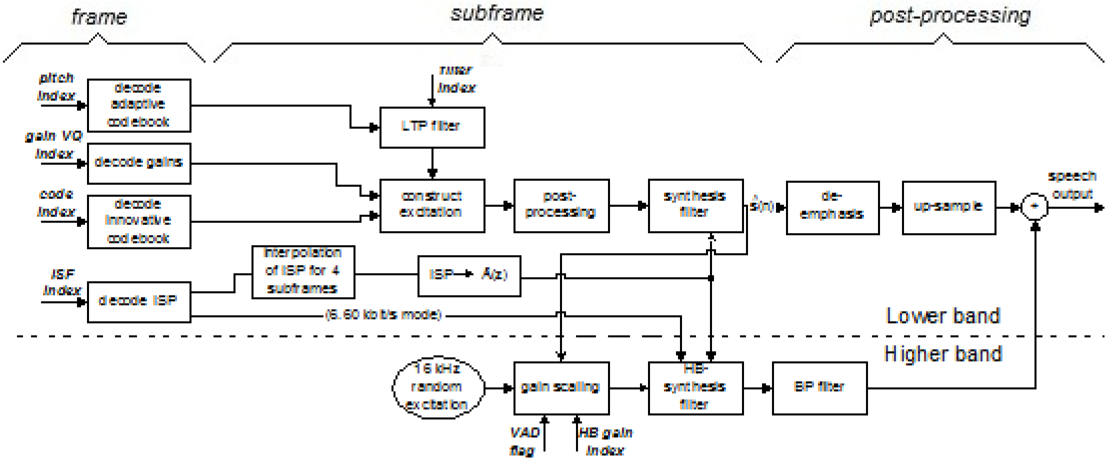


Figure 10:   Simplified block diagram of the adaptive multi-rate decoder [14],[32].

Decoding of the linear prediction (LP) filter parameters begin with the reconstructing of quantized immittance spectral pair (ISP) vectors from the received indices of ISP quantization. As in the encoder, the ISP coefficient update is done on every speech frame. After that the interpolation is done to obtain 4 interpolated vectors to compute a different LP filter at each sub frame [33]. Those vectors are converted to the LP filter coefficient domain for each sub frame to synthesize the reconstructed speech.

In every sub frame the adaptive codebook vector is decoded and found by interpolating the past excitation using a FIR type filter. The innovative and adaptive codebook gains are also decoded and jointly vector quantized. The speech reconstruction is computed on this stage and post processing is done before the actual speech synthesis.

The processing before the actual speech synthesis is involved into sub frame part of the decoder. One operation is adaptive gain control AGC, which removes the abnormal energy variation from the signal [33]. The other is anti-sparseness processing which is performed on the two lowest operating modes (8.85 and 6.60 kbit/s) because fixed codebook vectors offer such a low amount of information. This reduces the audible errors in the synthesized speech. A noise enhancer reduces the fluctuation in the energy in stationary signals, which increases the performance in stationary background noise.

For high frequencies from 6.4 to 7 kHz, an excitation is generated first to model the frequency range. The high frequency part is generated by filling the higher part of the spectrum with white noise, which is scaled in the excitation domain. After this the conversion to the speech domain is done by shaping the content with a filter derived from the same linear predicting (LP) synthesis filter used for synthesizing the downsampled signal. Before the speech signal is obtained the high-band speech is filtered with a LP and band pass filter from 6.4 to 7 kHz. Finally, the synthesized higher band signal is added to the lower band synthesized speech and the final output speech signal is completed.

5. MOBILE PHONE ACOUSTICS

In this chapter several factors that affect the quality of heard sound from mobile phone earpieces are presented. First, the processing of incoming audio from the speech decoder to the loudspeaker is introduced. After that the mechanical part is described by introducing the dynamic loudspeaker, and enclosures with leak types and reasons for certain mechanical selections are discussed.

5.1 Audio path from mobile phone's antenna to speaker

When a mobile phone antenna receives the speech signal from the sender, lots of signal processing is done before the audible sound from the loudspeaker. A rough description about the downlink path blocks from the speech decoder to the earpiece is shown in Figure 11. The important block is speech enhancements, which includes for example noise cancelling for removing the noise from received speech and multiband dynamic range controller (MDRC), which is described later in Figure 12.

The equalizer is needed correcting the magnitude response of the earpiece. If the loudspeaker cannot produce enough for instance low frequencies, the equalizer parameters can be tuned to increase the signal level on low frequencies. However, one disadvantage of this operation is increased distortion. The problem with an equalizer is that it fails to take the input signal level into account. This means that if the input signal is already loud before the equalizer, in the worse case, it is gained over the theoretical limits, which inevitable leads to noticeable distortion in the earpiece.

Upsampling from 16 kHz to 48 kHz is done to support a suitable sampling rate to the hardware audio codec. In the last block in the digital domain, the signal is converted from the digital to analog domain in the digital-to-analog (D/A) conversion block.

In the analog domain, the lowpass filter removes unwanted signal components before amplifications. Finally, the analog gain has to be adjusted to a suitable level for the earpiece. Before the earpiece a few passive components are added to protect the earpiece for instance from voltage peaks.

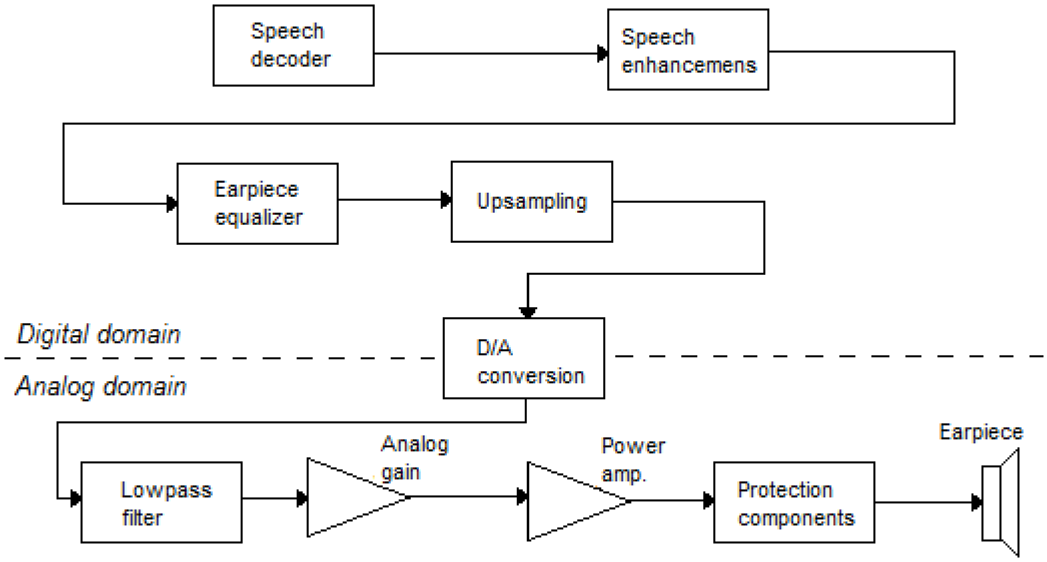


Figure 11: Simplified block diagram of the mobile phone earpiece downlink.

An important part of downlink audio path is the dynamic range controller DRC. A common feature of the DRC is that the input signal level is detected continuously, and according to this detection, a defined amplification to signal is performed [34]. The mentioned MDRC is an extension for the DRC, a device that divides the full frequency band into sub-bands, which are amplified separately. There are different ways to realize the DRC [34], but this thesis concentrates on the functional part of it. The basic principle is shown in Figure 12 and the meaning of different parts are explained below:

- *Signal deleted*: If the input signal level is very low the output signal is deleted by dropping the signal level for example -100 dBFS. Usually this kind of weak input signal level is noise and deleting it is natural to enhance SNR.

- *Expansion*: The level of quiet input sounds are increased and the dynamic range of the audio output signal is also increased. The steepness of this part is critical, because if the expansion is done too steep the low speech signal levels can be partly lowered resulting in audible errors in the earpiece. If the expansion is too mild, unwanted noise may be added to the output signal.

- *Amplification*: In this stage the input speech signal is amplified for instance 15 dB. The input speech signal is somehow normalized in the uplink and the amplification area is adjusted to be long enough to cover most of the speech signal.

- *Compression*: In simple terms, the loud sounds over a certain threshold are reduced, in this case the input signal limit is -30 dBFS. The operation decreases the dynamic range of the speech signal, but the advantage is that loud signal levels in a noisy environment are not amplified too much.

- *Limitation*: To prevent the loud output signal to reach the loudspeaker, the loudest output signal level is limited according to the loudspeaker performance. If the limitation is neglected, a considerable amount of distortion may be added to the output signal.

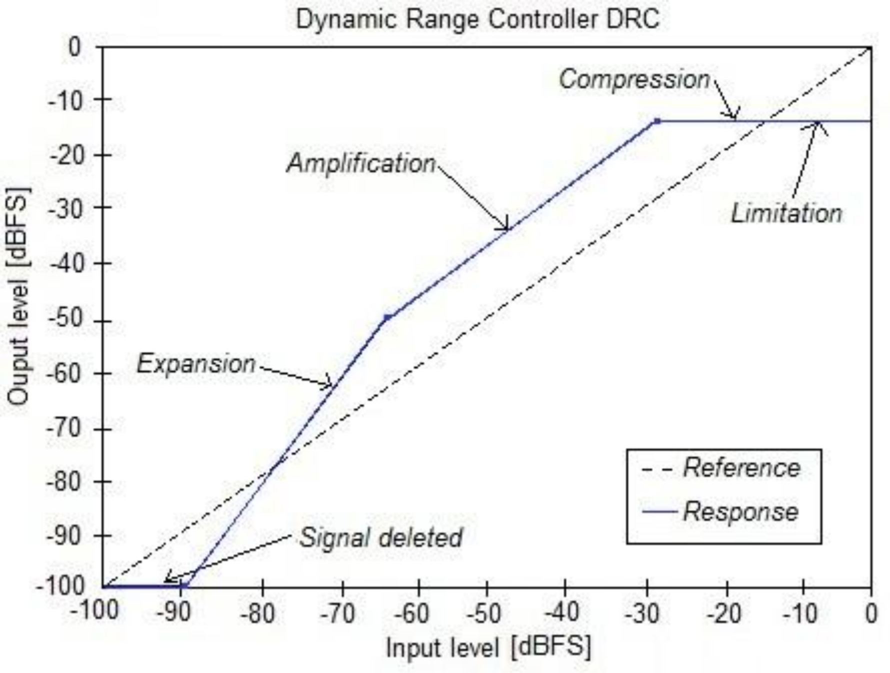


Figure 12: Dynamic audio compressor/expander level response [34].

The DRC was introduced in this section because even its performance is not shown in the frequency response and distortion measurements, it affects the end-user experience heard from the earpiece especially in *expansion* and *limitation* parts.

5.2 Audio module in mobile phones

The speaker plays an important role but acoustics matter as well. Phones with similar a earpiece can differ from each other due to different acoustics. The dynamic loudspeaker is introduced with different enclosures to give an idea of possible factors that affect the sound quality.

5.2.1 Dynamic loudspeaker

The dynamic loudspeaker is the most common speaker type in the loudspeaker industry. In practice, all speakers used so far in mobile phones are dynamic loudspeakers.

The basic idea of dynamic loudspeakers is to convert the electrical signal to an acoustical signal [18] and can be described as a four-pole model as in Figure 13. The input side has voltage and current, where as the output volume velocity and sound pressure. With the alternating current in a magnetic field the force tries to move the compact coil of wire. If the coil is attached to a large surface it moves the air more efficiently giving volume velocity to the air, which is heard as a sound.

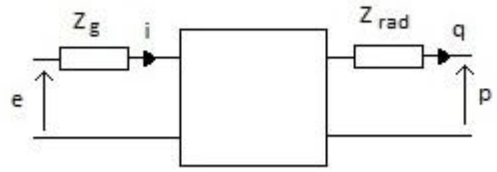


Figure 13: Four-pole network of the speaker [18]. Symbols are: e = input voltage, i = current, Z_g = impedance of electrical circuit, q = volume speed of oscillator, p = sound pressure, Z_{rad} = radiation impedance [18].

The current in a wire in a magnetic field produces a force on the wire. If a single wire is moving in a uniform magnetic field it represents the simplest coil transducer. The coil experiences force in the axial direction and the total force is:

$$F_{mag} = Bli, \tag{26}$$

where F_{mag} is the force produced by a current i [N], B is a magnetic-flux density in tesla [T], i is the alternating current in amperes [A] and l describes the length of wire in the magnetic field [m].

The moving coil loudspeaker system can be presented with an acoustical equivalent circuit described by Hall [28]. By taking a closer look at Figure 14, the speaker is described by an electro-mechano-acoustical circuit. The coil has electrical resistance R_e and inductance L_e. The amplifier with output impedance Z_g supplies voltage E_g which drives current i_g through the coil. This causes force $F=Bli$ on the cone and coil, with resulting motion v. The cone and coil are considered as a mechanical system with a mass m and mounting stiffness C_m , including flexing of the material resistance R_m. After that, the volume velocity $U=vA$, where v is the mechanical velocity of the membrane and A is cone area *A*. Volume velocity *U* works against the radiation impedance $Z_{a,rad}$ of the surrounding air to generate sound pressure *p*.

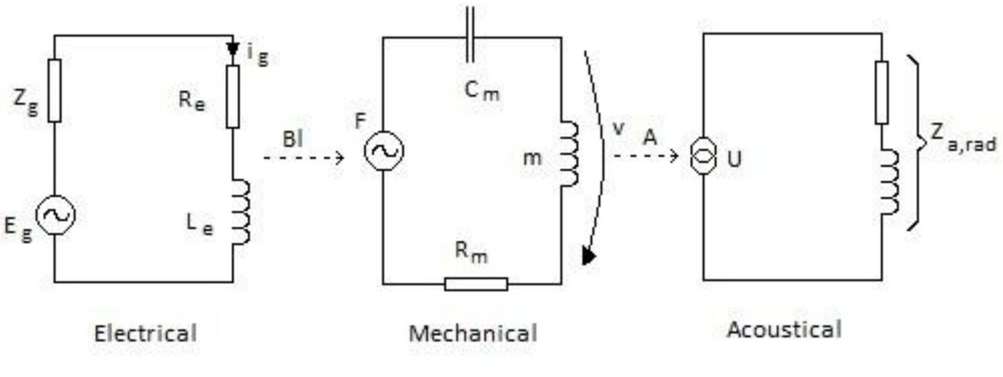


Figure 14: Electro-mechano-acoustical circuit of moving coil loudspeaker system. The symbols are explained above the figure in the text [28].

The structure of a dynamic loudspeaker is quite simple. Usually the wire is wrapped as a coil and situated between the magnetic poles. The purpose is to maximize the length of the wire where the magnetic field is constant and perpendicular to the wire. As mentioned earlier neither the coil itself does move much air, nor produce sound. The efficiency is increased by attaching the coil to a movable, light and relatively large surface diaphragm which carries lots of air along with it. Low frequencies generally need a larger radiating area, whereas high frequencies are produced with a smaller area. In larger speakers, or home stereo speakers, the diaphragm is usually a cone, which is fairly stiff and light. The diaphragms in small speakers used in a mobile phones are also quite stiff even though the material is thin.

By looking at Figure 15 the main parts [27] of the dynamic loudspeaker are shown in a cross-section picture. However, exactly this kind of shape is not used in mobile phone earpiece, but the principle is the same.

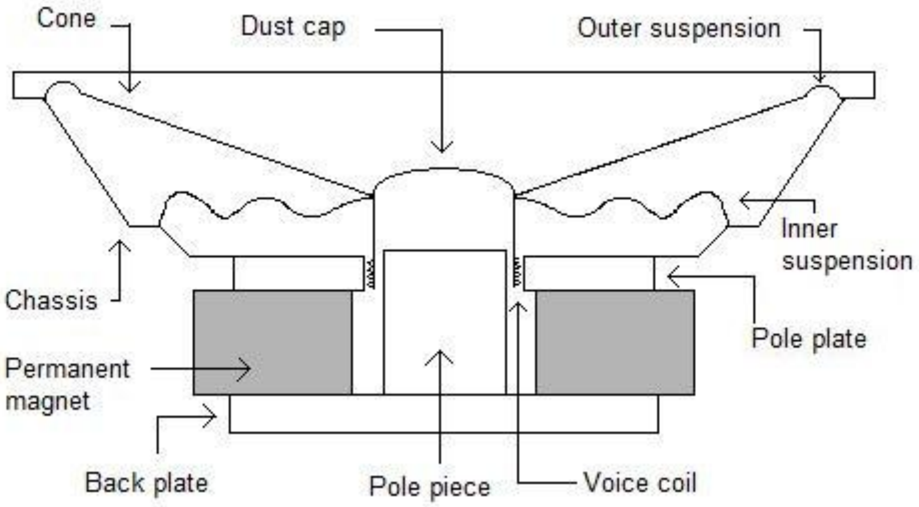


Figure 15: Cross-section sketch of a dynamic loudspeaker [27].

To minimize the non-linear response, the permanent magnet must be selected to produce a constant field and the voice coil should always have the same amount of turns in every displacement inside the gap between the poles [29]. The movement area of the voice coil has to be limited in the constant field area, because if a coil exceeds that area the magnetic force *Bl* drops causing non-linearity. Another problem occurs if the speaker diaphragm movement distance is too long and exceeds the linear operation area, which may cause mechanical damage to the speaker. In order to reduce impedance variation, the chassis is made either to be able to conduct the heat away from the voice coil or the chassis endures high temperatures and this way the voice coil stays in a stable condition even in harder usage.

Voice coil suspensions keep the speaker membrane accurately centered in the magnet gap, which is important enabling only the axial direction movement. One other function of the suspension is that, when there is no signal, the membrane is returned back to the equilibrium position.

The main reasons for using a dynamic speaker are the low operational voltage, small size and low price. There are some drawbacks like low efficiency (usually 1%) and frequency response, which is poor at low frequencies due small effective radiating area and short membrane movement distance. The simulation of enclosure and speaker element for design purpose is fairly straightforward due to the long history of dynamic transducer studies.

5.2.2 Leak types, front and back cavity

A loudspeaker without an enclosure design does not provide very good acoustic performance. Usually there are lots of compromises in mobile phone acoustic design due purpose of use or mechanical design. The main purpose of mobile phone earpiece is to reproduce speech, which has been narrowband until these days. In upcoming years, there will be a need for wideband capability and that must be taken into account when designing the acoustics for mobile phones. Therefore the front and back cavities with different leak types are introduced in this section.

The front of the loudspeaker

There are two main possibilities to realize the front part of the speaker, called the front resonator and open front.

Front resonator: The front cavity usually consists of the cavity itself and a cover with sound holes. The purpose of the front cavity is to boost high frequencies thus reduce the need for equalization. The other function is to provide protection against dust, water and other external damage, which could be harmful for the speaker without the front cavity. The disadvantage of the front cavity together with the sound holes is that it is a new source of tolerance errors, but proper front cavity design can decrease the amount of tolerance effects. Also, the cavity requires space, which is not available adequately in mobile phone.

Open front: If there is no space for a front cavity, the speaker can be placed so near the phone cover that the cavity is very small or does not exist. This way the acoustic resonances are shifted to higher frequencies above the speech band, but the boosting effect of the front resonator for the higher speech frequencies in the usable band is lost. This leads to much heavier DSP equalization. In addition, the open-front design does not include a lowpass feature, which can be used to filter out unwanted for example radio frequency (RF) buzzing noise just above the speech signal band. Both of the explained implements are shown in Figure 16.

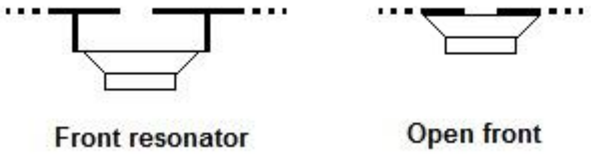


Figure 16: Different mobile phone earpiece front realizations [31].

Back side of the loudspeaker

There are four main design and various hybrids available for designers to choose for the back side of the loudspeaker.

Open back: The back of the loudspeaker is left open to the space of air inside the phone. This method is the most common in all earpieces [31]. Even if the sound from the back of the loudspeaker is routed through the PWB behind the speaker, the realization is called open back. The advantages are ease of design and small space consumption. Problems can occur on low frequencies if a large external leak is included in the mechanics.

Closed back: In this case the loudspeaker has its back enclosed in a cavity that is sealed or contains a small acoustically damped leak. The leak is only connected to ambient air or air inside the phone, not to the ear. The good side of this is the isolation to the microphone though the air path inside the phone. The downside is that the cavity should be large, around a few cm^3. In practice, the open back realization is preferred for its small space occupation offering almost as good a result as the closed back.

Vented: The idea in vented back enclosure is to work as a bass booster. The earpiece has a back cavity that is hermetically sealed apart from an opening with a defined cross-sectional area and length (pipe) behind the loudspeaker. The gained resonator is tuned near the lower limit of the frequency range of the earpiece. To make the vented structure to work, the outer end of the vent has to be routed directly or indirectly to the user's ear. If this is neglected and the vent is left inside the phone without acoustical connection to ear, the performance will be worse than with other implementations. The reason for using this design is the boosting effect for low frequencies with lower distortion thus it is well suited for a small speaker in wideband designs. The problem with this design is the same as with closed back design, which is the required large cavity.

Tube-loaded: The realization is about the same as vented construction, but this case the cavity is much smaller and the vent is narrower and longer. The advantage of this method is small bass boost, which is important in wideband implementations. Even a relative small speaker with stiff suspension can reach low frequencies but the speaker has to handle the required higher displacement.

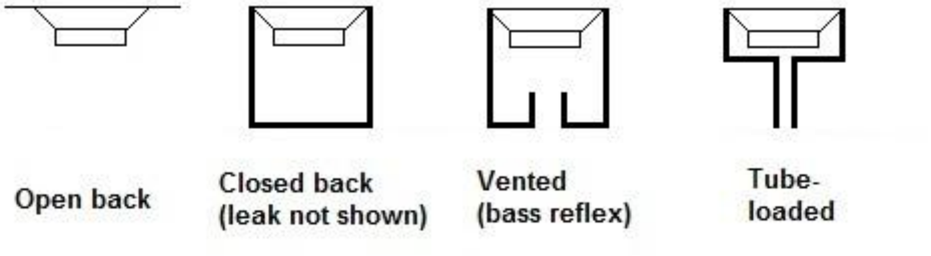


Figure 17: Four different earpiece enclosure back types used in mobile phone [31].

Leak types

Basically there are three different types of leak to put into practice shown in Figure 18.

No leak: When the phone is held against the ear, only the natural leak between the ear and the surface of the phone cover is present. This kind of leak is the simplest of all realizations having high leak tolerance, which means that the sound of the earpiece is relatively insensitive to variations in the leak between the phone and the ear.

External leak: This case an intentional acoustic leak lets some of the sound pressure escape from the ear to the ambient air outside. The leaking occurs even if the phone is sealed against the ear. This type of design is common and works well increasing further leak tolerance. Also the tuning is easier due the leak.

Internal leak: The idea is the same as the external leak, except the leak is going from the front cavity to the ambient air. Usually this realization performs worse than the external leak implementation due to equalization for high frequencies. The internal leak option is not recommended, except if the earpiece and integrated hands-free (IHF) speaker have to be combined or lack of space prevents other leaks in the phone cover.

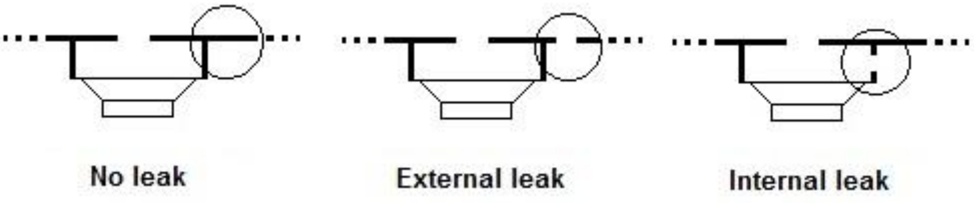


Figure 18: Different leak types used in mobile phone earpiece enclosure [31].

5.2.3 Loudspeaker implementation in different phone models

As presented earlier in Chapter 5, good performance depends on many things. There are many rival parameters affecting the size of the loudspeaker and enclosure selections. The planned phone price defines quite much for example the components that there will be in the phone. Mechanical design limits or allows modifications to acoustics as well. Some of these factors are listed below:

- Narrowband or wideband phone
- Phone price

- Dust and water protection
- Mechanical design
- Designer's set of parameters

If the phone is a wideband model the requirement for earpiece sound production performance is different to narrowband. Using the small speaker gives more space for other components in the phone and can be cheaper, but the low frequencies on wideband cannot be reproduced as purely, or at all, as with a larger speaker. The mechanical design containing cavities and leaks described earlier with DSP may help, but a speaker has its limits and cannot break the physical laws. If the speaker is large, lousy mechanical design or audio designers tuning parameters may be ruining the advantages that could have been achieved by the speaker. On the other hand, using a large speaker gives more margin to audio designers compared to small speakers.

6. OBJECTIVE MEASUREMENTS

It is important to show the objective results, when the subjective results are analyzed. The purpose of objective measurements is to find out if there is any distinct differences between the speakers 1) without the phones, 2) integrated to the phones without audio processing and 3) integrated to the phones with audio processing. First, the theory of objective measurements is presented and then the measurements with results of the used phones and speakers are shown in this chapter.

6.1 Theory of audio measurements

Before the measurement results the theory behind the objective measurements are presented in the next three sections. Impulse response, frequency response and distortion methods are described to help to understand the measurement results later in this chapter.

6.1.1 Transfer function

The transfer function describes an ideal system behavior with any kind of stimulus. An ideal physical system has four properties [21]:

1) *The system can be physically realized* means that the system cannot produce an output before input is applied.

2) *Constant of its parameters* when the system is time invariant thus the response of the system is constant for all time values.

3) *Stability* limits the system's output to be a finite signal for a finite input signal.

4) *A Linear* system is additive and homogeneous. If the output signals are $y_1$and y_2 with input signals x_1 and x_2. *An additive system* produces a summed output $y_1 + y_2$ from summed input x_1+ x_2. *A homogeneous* output $cy_1$is produced from input cx_1, where *c* is a random constant.

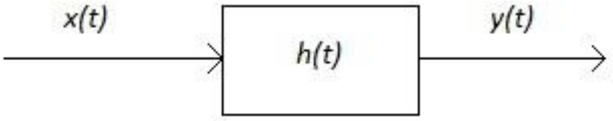


Figure 19: Linear, ideal system h(t) with one input x(t) and output y(t) signals [21].

The unit impulse function is defined as follows:

$$h(t) = y(t)$$
$$when$$
$$x(t) = \delta(t)$$

Where *h(t)* is unit response function of the system, y*(t)* describes the output of the system, *x(t)* input of the system and *δ(t)* is the ideal impulse, i.e. delta function, *t* is the time from the moment when delta function enters the system. The duration of ideal impulse is defined to approach to zero, so, in other words, the duration is infinitely short. Also, the amplitude and energy is defined to be infinity and the integral equals 1 [22]. If the ideal impulse would be used for measuring the speakers of phones in this thesis, the results with good signal-to-noise ratio would contain a high amount of distortion, or even worse, break the speakers. It is obvious that these kind of parameters are not for the practical part of this thesis but only for theory.

Instead of using the theoretical and unpractical method mentioned earlier, sine sweep is used to measure the frequency response of the speakers. The simplest form of sine swept frequency measurement is linearly or logarithmically variable sine wave, which is fed to the device under measurement, DUT. Output signal represents the magnitude response of the device.

Because the sinusoidal stimulus contains energy concentrated instantaneously at one frequency, noise and other artifacts disturbing the measurement can be filtered away by using tracking filter, i.e. a narrowband bandpass filter. One advantage of sine sweep is its ability to measure simultaneously frequency response and the non-linear distortion [36]. According to Farina in [36], from a sine signal used in an exponentially varied frequency, it is possible to deconvolve simultaneously the linear impulse response of the system and separate impulse responses for each harmonic distortion order.

6.1.2 Frequency response

When the system's output spectrum in response to an input signal is of interest, the frequency response is the right measure. It often helps designers to implement systems by offering additional information about the system behavior. The characteristics of the system can be illustrated by a linear transform of the unit impulse response. The transform function of a system can be represented in the Laplace form of the impulse response.

$$H(s) = \int_0^{\infty} h(t)e^{-st}dt \tag{27}$$

where *H(s)* is the complex-valued transfer function of a system and $s = \sigma + j\omega$ describes the complex frequency variable. The Laplace function, like the impulse response is defined to begin at *t* = 0.

The most powerful method to represent the response in the frequency domain is the Fourier transform [21] of the impulse response.

$$H(f) = \int_0^{\infty} h(t)e^{-j2\pi ft}\, dt \tag{28}$$

where *H(f)* is the complex frequency response of the system and *f* is the real-valued frequency. When $\sigma = 0$ the Fourier transform (28) becomes the transfer function (27) and it is investigated on the $j\omega$-axis. The transfer function is not very practical and, therefore, the frequency function is used instead. Usually, the absolute value of the frequency response called the magnitude response $|H(f)|$ describes the frequency response curves.

The transform domain representation can be inversed to the time domain by inverse Laplace in the s-domain or Fourier in the f-domain transform.

$$h(t) = \frac{1}{j2\pi}\int_{\sigma-\infty}^{\sigma+\infty} H(s)e^{st}ds \tag{29}$$

$$h(t) = \int_{-\infty}^{\infty} H(f)\, e^{j2\pi ft}df \tag{30}$$

where σ is the real part of the complex frequency variable s.

6.1.3 Distortion

If a speaker with a completely linear transfer function would produce an identical output signal compared to input signal there could not be any distortion. It is not a surprise that a small speaker in mobile phone, capable of producing high sound pressure level cannot play undistorted sound. The nonlinear distortion produces unwanted signal components, which does not exist in original signal and adds them to the output signal.

The harmonic distortion appears on a clean sine sound in harmonic components. If the spectrum of the signal consists of fundamental frequency A(1) and the amplitude of i^{th} harmonic component *A(i),* the total harmonic distortion (THD) is defined [1] as

$$d = 100\% \frac{\sqrt{\sum_{i=2}^{N} A(i)^2}}{\sqrt{\sum_{i=1}^{N} A(i)^2}} \tag{31}$$

where d is harmonic distortion, $A(i)$ is amplitude of i^{th} harmonic component and N is the number of harmonic components.

The distortion measurements in this thesis are done with all non-harmonic components, background noise and noise from measurement equipment. This kind of distortion is called THD+N. Not all distortion is a bad thing, mentioned in [1] that low order harmonics can make the speech sound more pleasant in a telephone line than undistorted speech.

6.2 Earpiece measurements in free field

The phones under measurements and the listening test have different size of speakers. The physical measurements of the speakers are shown in Appendix A: and Appendix B:. Those two speakers were measured in free field to obtain information about the speaker performance before integrating it to the mobile phone. Both of the earpieces were measured by Lauri Veko in the anechoic room (AR1) at the Salo Nokia premises in November 2007.

6.2.1 Earpiece measurement procedure and equipment

The earpieces that were under evaluation were measured in the International Electrotechnical Commission (IEC) standard baffle (a plate with a hole for speaker). The used measurement setup was the following: The measurement adapters for both speakers were free air adapters and designed for those two speakers. Basically, the adapter had a hole in the front part, where the component sound hole was and the back part was open. The measuring distance between the speaker and 1/4" free field microphone was 1 cm. The whole measurement setup is shown in Figure 20. The measuring equipment is listed in Table 2.

Table 2: Test equipment used in measurements of the earpieces in a baffle.

Instrument	Type	Comment
Audio measurement system	Audio Precision System Two Cascade	Control software ApWin

Instrument	Type	Comment
Audio Precision	Audio Precision 2700 series	Amos XG controls Audio Precision, ApWin
Condensator microphone + preamplifier	B&K Type 4939 B&K Type 2670	1/4" mic + its preamp
microphone amplifier	B&K Nexus	Mic amplifier
Impedance box	1ohm Shunt resistor	
Rotel amplifier	amplifier with fixed gain +6dB	Speaker amplifier

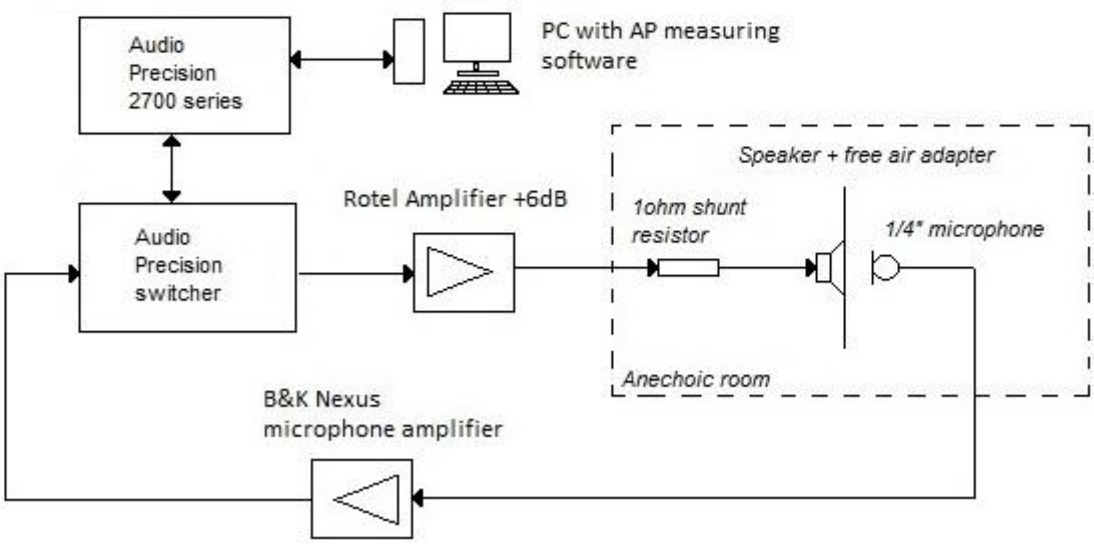

Audio Precision 2700 series
PC with AP measuring software
Speaker + free air adapter
Rotel Amplifier +6dB
1ohm shunt resistor
1/4" microphone
Audio Precision switcher
Anechoic room
B&K Nexus microphone amplifier

Figure 20: Speaker measurement set-up. A PC controls the Audio analyzer, which brings the measurement data back to the computer.

6.2.2 Results of earpiece measurements

The speakers under evaluation were measured in a frequency range from 200 Hz to 20 kHz. The results of the frequency response measurement can be seen in Figure 21 and distortion in Figure 22. The input voltage for both measurement was set to 0.179 Vrms.

Frequency response of the earpieces

If the wideband codec frequency range 50 - 7000 Hz is examined from the Figure 21, the high frequency area near 7 kHz is about the same for both speakers. On the other hand, at low frequencies, the small speaker performance is clearly worse than for the large one. In fact, the frequencies under 400 Hz are 10 dB more silent on the small speaker, which is not possible to compensate in the mechanical design and, therefore, is seen later in *earpiece integrated to phone* in Section 6.3 results.

As can be seen from Figure 21 the large speaker frequency response curve -3 dB point is around 300 Hz, when the small speaker has it at about 500 Hz. Also, the frequencies between the two resonance peaks (big speaker 400 Hz-5000 Hz, small 600 Hz-7000 Hz) varies in the large speaker only 4 dB and the small one 7 dB.

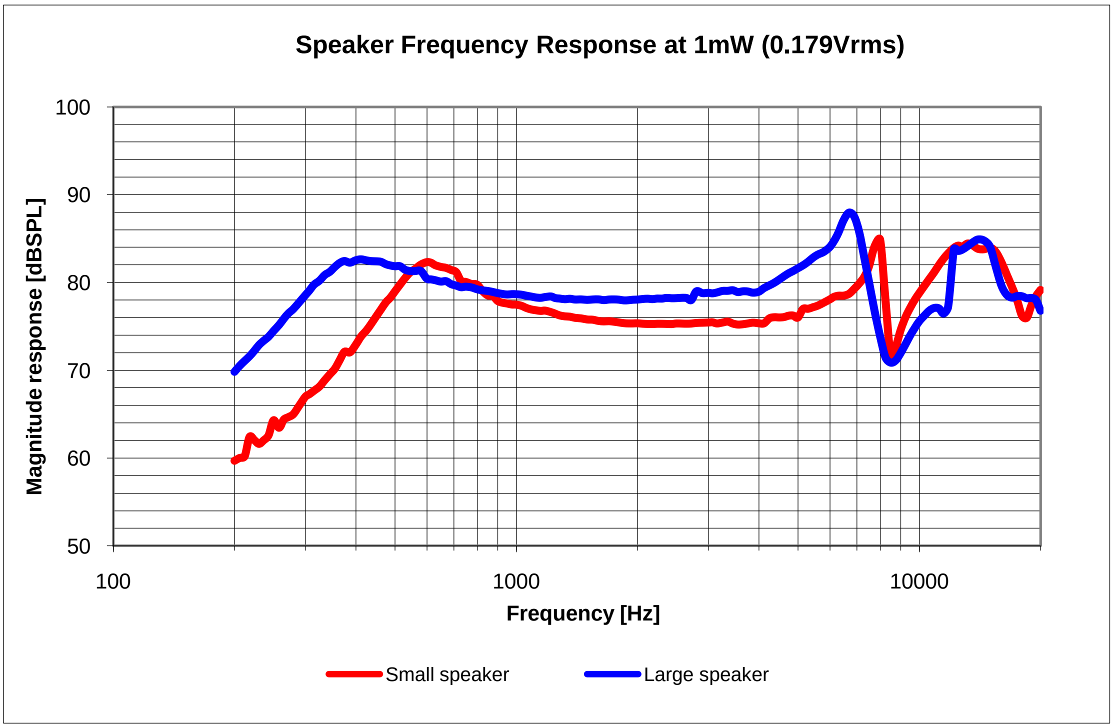


Figure 21: Small and large speaker frequency response in frequency range 200 Hz - 20 kHz measured with free air adapter.

Distortion in earpieces in free field

Distortion on both speaker is plotted in the same Figure 22 and notable thing is that background noise is also included in the results. However, the small speaker has more distortion below 650 Hz ending up to 11%. The frequency band is from 200 Hz to 20 kHz.

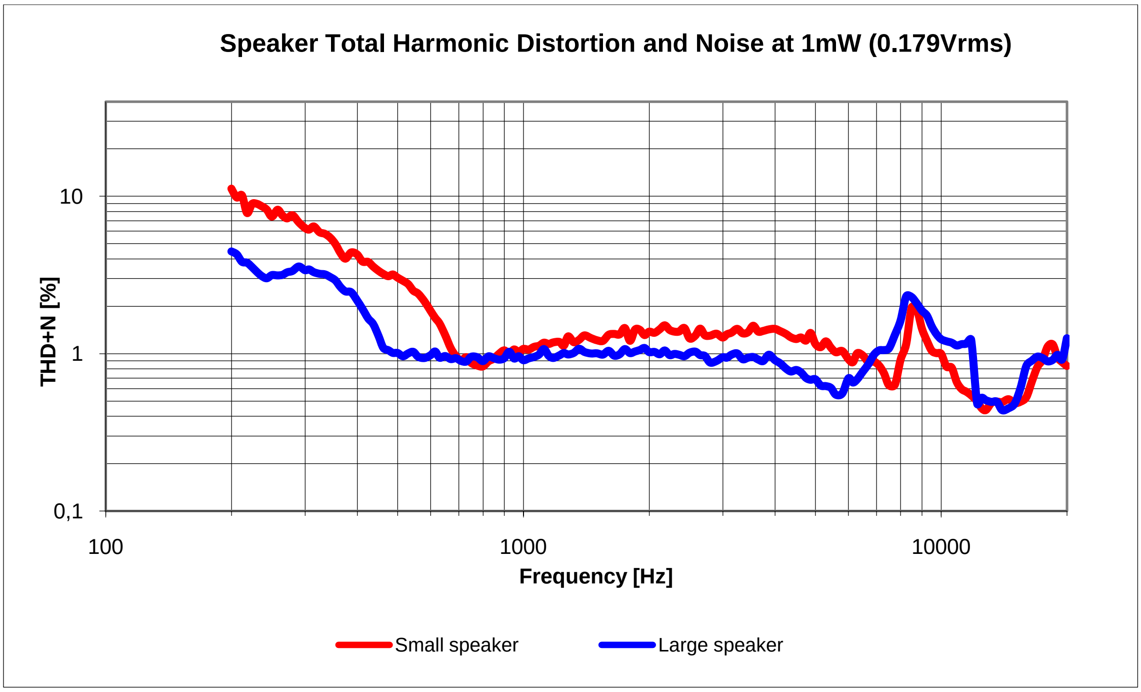


Figure 22: Small and large earpiece total harmonic distortion and noise in frequency range 200 Hz - 20 kHz measured with free air adapter.

6.3 Measurement for wired earpieces integrated to Nokia mobile phone

After the free field measurements in Section 6.2, the speakers' performances can be compared more or less in a theoretical aspect. If a speaker is measured in a free field adapter it does not sound the same way when placed in a proper enclosure. Measuring the earpieces wired to the phone means that the earpieces are integrated to the real phone without the audio signal processing. Using the HATS' ear for measurement, the results are one step closer to the end-user experience.

6.3.1 Selection criteria's for phones in the test

The phones that are measured in the next sections are introduced shortly before the measurement procedure. Both of the phones were selected for this thesis on three criterias: 1) Phones have different sized speakers 2) Support for AMR-WB codec available, 3) is available on the market thus is not just a prototype. The other phone is a Nokia 6220 classic introduced in Q208. It has the small speaker measured in Section 6.2. The other is the wideband phone, the Nokia 6720 classic, introduced in Q209. Furthermore, it contains the large speaker.

6.3.2 Measurement equipment and procedure for wired earpieces

The idea of this measurement is to get data about the speakers integrated to the phone without the influence of audio signal processing. The phones were positioned to HATS according to the designers directions (Appendix F: and Appendix G:) and the phone earpieces were wired directly to Audio Precision through a B&K power amplifier. After the HATS ear the microphone signal was amplified in the B&K microphone multiplexer and connected to a PC through Audio Precision. A block diagram of the measurement is shown in Figure 23.

Table 3: Used measurement equipment in HATS measurement by wiring the earpieces directly to the power amplifier.

Instrument	Type	Comment
Audio measurement system	Audio Precision switcher Type SWR 2122	Control software ApWin
Audio Precision	Audio Precision 2700 series	Amos XG controls Audio Precision, ApWin
Power amplifier	B&K Type WR1105	Loudspeaker amplifier
Microphone multiplexer	B&K Type 2822	Microphone amplifier

Before the actual measurement, the phone covers were opened and wires were soldered to the earpiece springs. The proper soldering was confirmed with a multimeter by measuring the speakers' resistances, which were about 30 Ω.

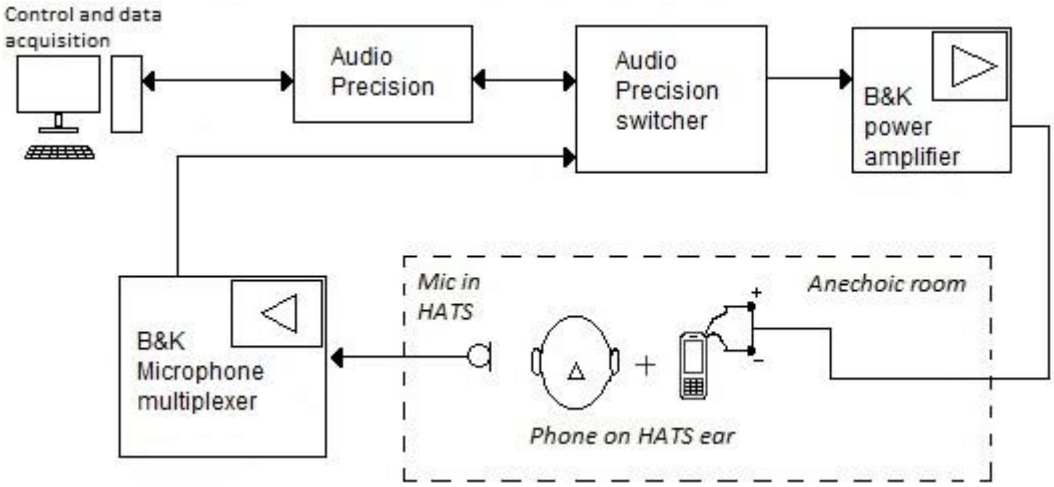


Figure 23: Measurement setup for phones measured on HATS by wiring the earpieces.

The voltages used in the measurement were defined by Audio Precision and are presented in Table 4.

Table 4: The input voltages for phone earpieces measured on HATS.

Type	Input level [dBV]	Input level [mVrms]
ERP	-20	114
DRP	-20	114
DRP	-15	202.7
DRP	-10	360.5
DRP	-5	641.1

6.3.3 *Results of the wired earpiece measurements on HATS*

After the measurements Audio Precision plotted the results to Excel. The frequency response and distortion results are shown below.

Frequency response

When phone measurements on HATS are plotted for different voltages, it can be seen that the speaker performance is noticeably different, especially at lower frequencies. There is a possibility to measure frequency response with ear-drum reference point (DRP) or without ear reference point (ERP) the influence of HATS' ear auditory canal. The reason for using DRP is that ERP measurement does not contain distortion measurement in Audio Precision due to unlinear distortion at different frequencies. Filtering the unlinear distortion would not be such a successful process. However, the behavior of the speaker integrated to the phone is clearly shown in the DRP measurements.

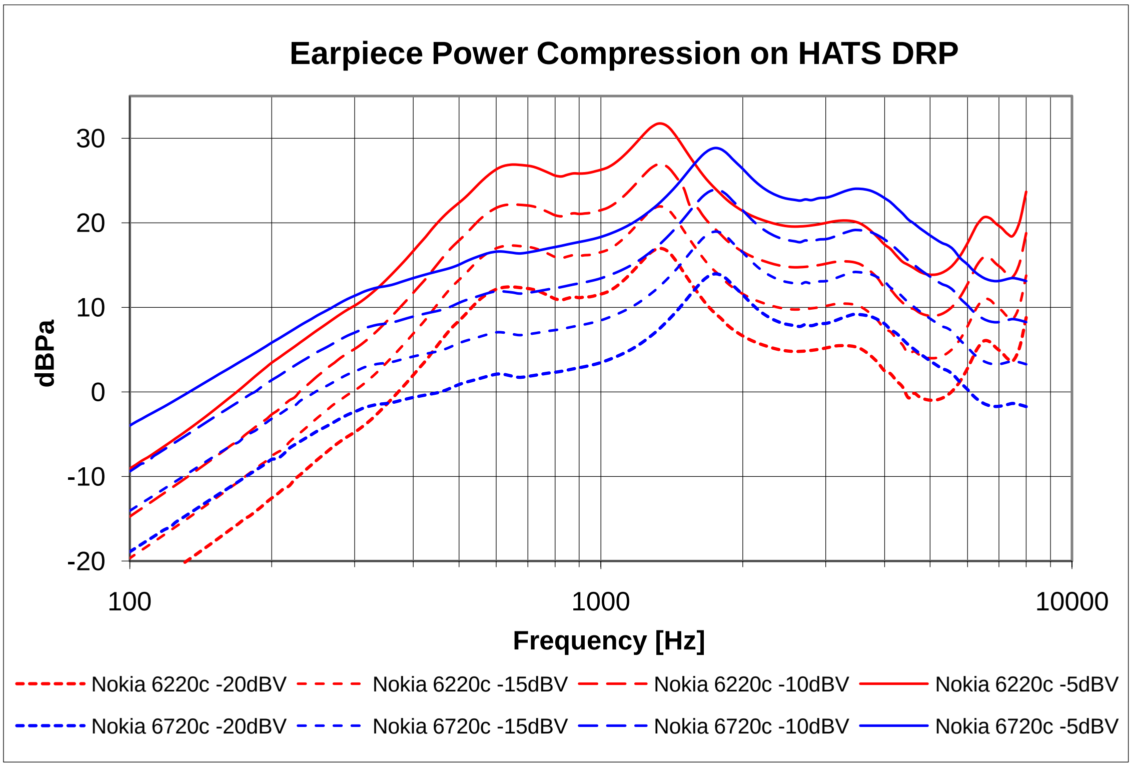


Figure 24: Frequency responses on range 100 - 8000 Hz measured on HATS using different input voltage levels in N6720c and N6220c speakers.

The Ear Reference Point (ERP) results are shown for comparison for full audio path measurements in Section 6.4. The Nokia 6220c is sealed on the HATS ear better than the Nokia 6720c, which can be seen in Figure 25 at frequencies lower than 1 kHz. The boosting effect from sealing helps the designer's tuning work to get the phone to fit into the 3GPP frequency response mask. The descent of the frequency response of the Nokia 6220c from 600 Hz to 100 Hz is steeper than for the Nokia 6720c.

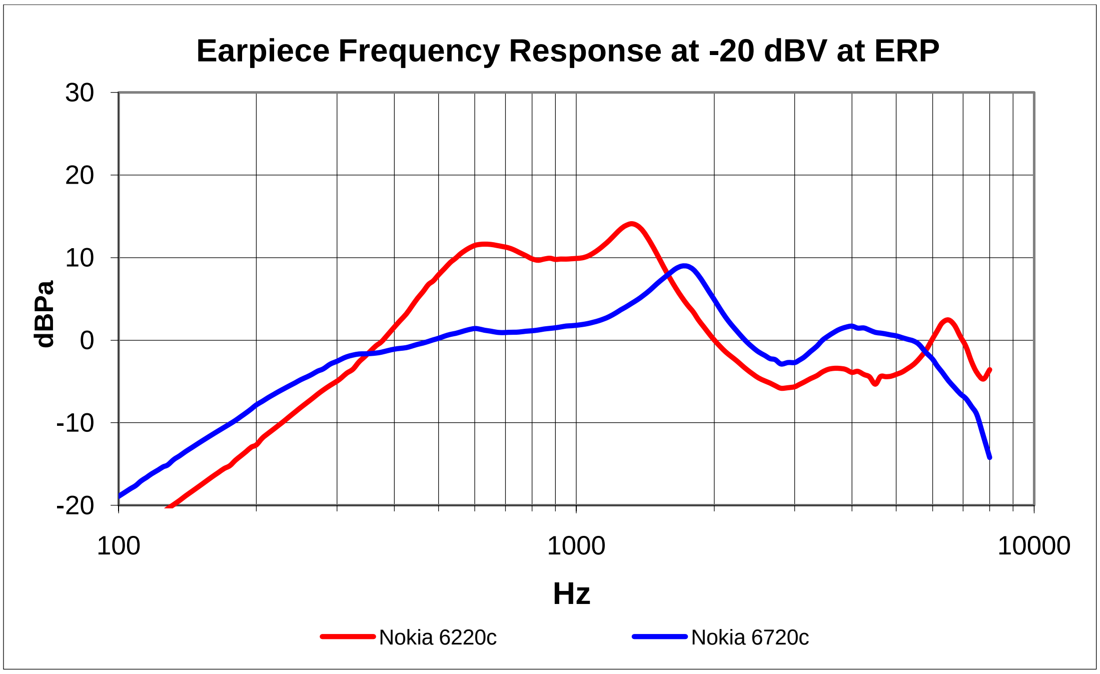


Figure 25: Phones measured on HATS ERP position on frequency range 100 - 8000 Hz. HATS' ear effect is filtered away by Audio Precision measurement system.

THD+N

In the earpiece total harmonic distortion and noise (THD+N) measurement on different voltages, the small speaker performs considerably poorly compared to the large speaker. The distortion begins to rise faster than the large speaker under frequencies of 700 Hz. The frequencies under 450 Hz for the small speaker THD+N results are inaccurate due the incapability of reproducing such low frequencies. Basically, the membrane is so small that sound production is impossible at those frequencies.

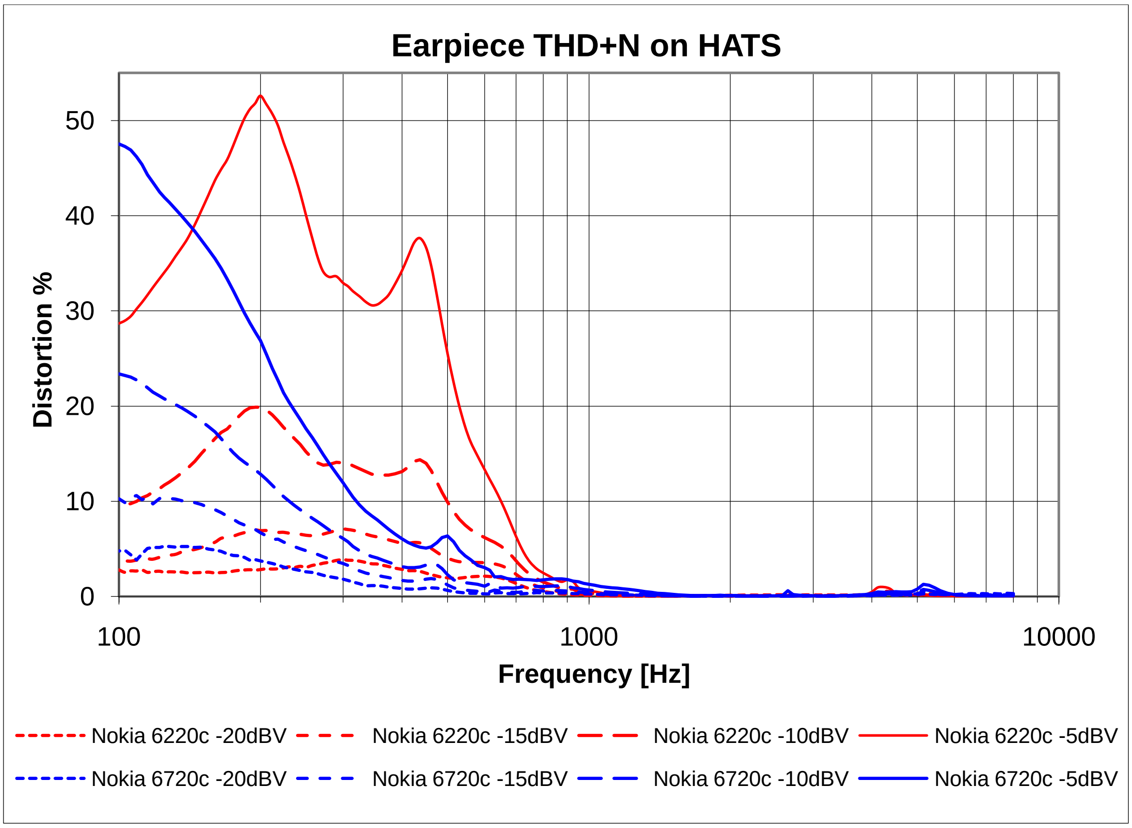


Figure 26: Nokia 6720c and 6220c THD+N on different earpiece input voltages on HATS. Measured frequency range is 100 - 8000 Hz.

6.4 Nokia mobile phone measurements over the air

The objective 3GGP [26] measurements of two Nokia phone models are shown in this section. The idea is to compare over the air results to the speaker measurements done in Section 6.2. It is also important to get information about the speaker integration and audio path effects in the measurement results. These results describe the end user hearing experience during a call.

6.4.1 Information about the phones in the test

All mobile phones that are on the market have been measured in a type approval test. For this test, the designer defines the nominal volume level, which is a volume level between 2/10-9/10. The selected nominal volume levels for both phones are shown in Table 5. These volumes were used because the phones have been tuned to fit to the masks on these specific levels.

Table 5: Nominal volumes of two Nokia phones.

	Nokia 6220c narrowband	Nokia 6220c wideband	Nokia 6720c narrowband	Nokia 6720c wideband
Nominal volume	5	4	6	6

The AMR-WB speech codec was enabled in both phones for the tests.

6.4.2 Measurement equipment and procedure

Both of the phones were measured on HATS with 3GPP specification release 7 [26]. 3GPP specifies test methods to allow the minimum performance requirements for the acoustic characteristics of GSM and 3G terminals for both narrow and wideband. The used measurement equipment in 3GPP measurements were proceeded with the list of instruments in Table 6. This is the only measurement where Audio Precision was replaced by Audio analyzer UPL-16.

Table 6: Equipment used in 3GPP measurements.

Instrument	Type	Comment
Radio communication tester	Rohde & Schwarz CMU 200	GSM & WCDMA network used
Rohde & Schwarz UPL-16	Audio analyzer	Sents result data to PC Amos XG controls it
Head And Torso Simulator	B&K	Includes artificial ear with microphone
microphone amplifier	B&K Nexus	HATS mic amplifier +20dB

Once the phone was positioned to HATS the measurement was started. A PC controlled the UPL, which fed the measurement signal to the radio communication tester and the phone earpiece speaker signal was measured on the HATS' ear. Finally, the recorded signal from the HATS ear is amplified 20 dB with B&K Nexus. The procedure is shown in Figure 27.

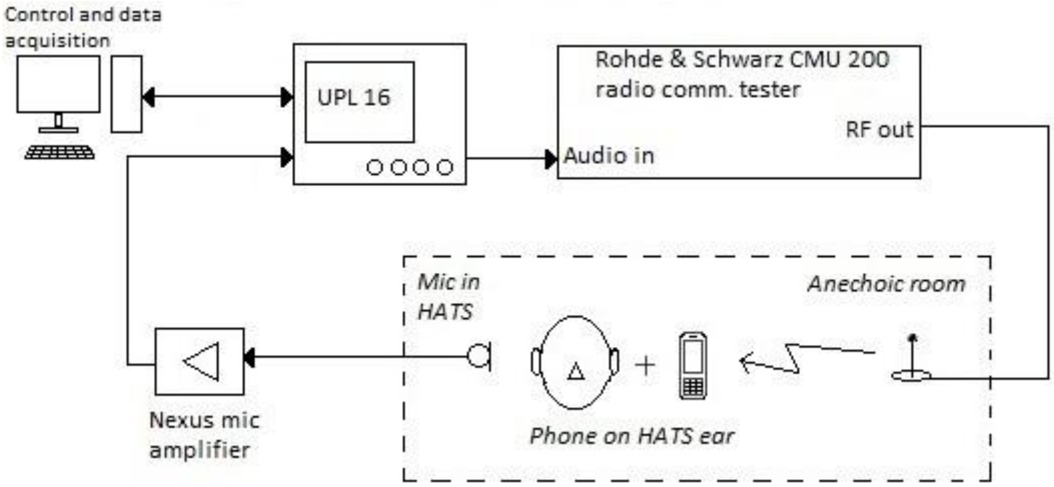


Figure 27: Measurement setup for phone speaker frequency response and distortion.

A picture of HATS used in this measurement and later in Section 7.2 for recording the files for subjective test is shown in Figure 28.

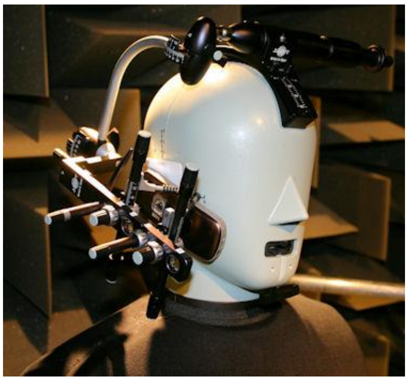


Figure 28: Nokia 6720 classic positioned to 3GPP measurements HATS in anechoic chamber.

Network settings for both phones were as in Table 7.

Table 7: CMU network settings in 3GPP HATS measurement.

GSM		WCDMA	
PCL	5	Uplink	1852.4 MHz
TCH	35	Downlink	1932.4 MHz
Codec mode	12.20 kbits	Codec mode	12.65 kbps
Speech codec	low	Speech codec	low

6.4.3 Measurement results

To ensure that the phone to be used in the recordings for subjective test is not a faulty one, five phones of each model were measured. An average phone was selected based on frequency response and distortion measurements. Differences in results between the measured phones results were mostly inside ±1 dB.

Frequency response

The frequency response follows the results in the previous Section 6.3.3. The Nokia 6720c has a flat response, while the Nokia 6220c fails to stay within the limits on narrowband (red dash line in Figure 29).

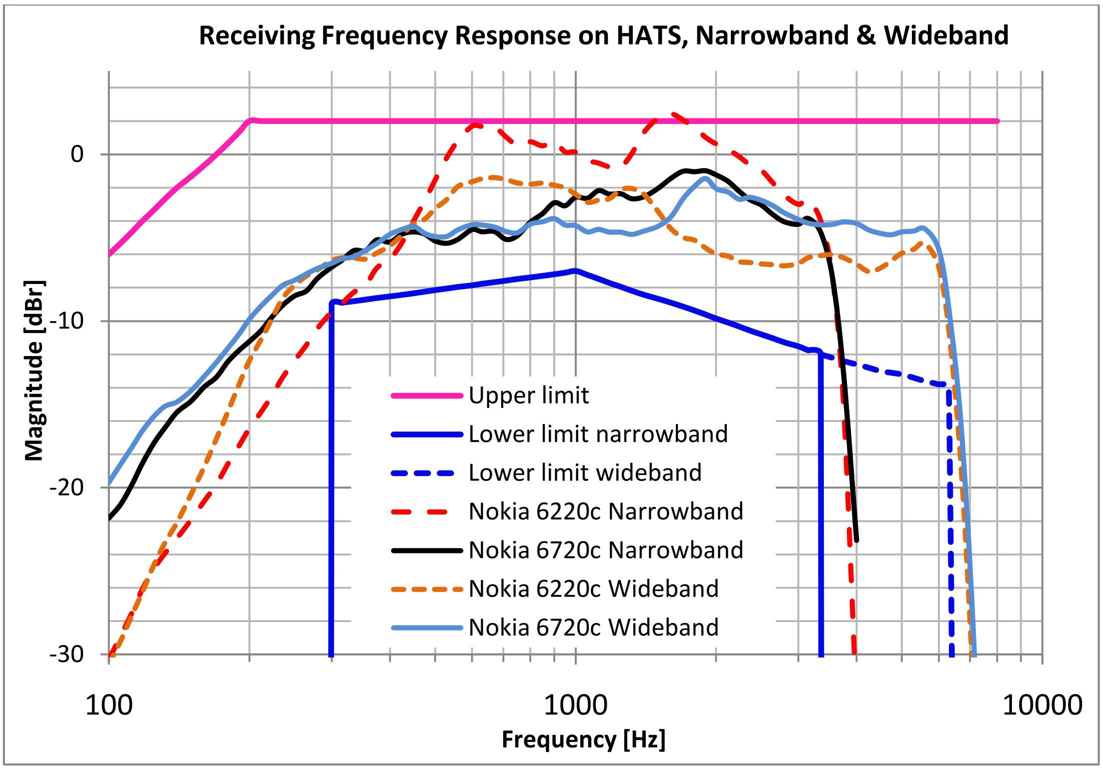


Figure 29: Results from narrow and wideband earpiece frequency response measurements of two Nokia phones on nominal volumes. Lower and upper limits are solid line for narrowband, wideband lower limit is marked with the dash line. The limits are from 3GPP version release 7. Narrowband bit rate was 12.20 kbit/s whereas on wideband it was 12.65 kbit/s.

Distortion

Distortion is presented as separate figures for narrowband and wideband to clarify the plotting. The Nokia 6220c performs worse on both narrow and wideband at volume levels 4/10 and 5/10.

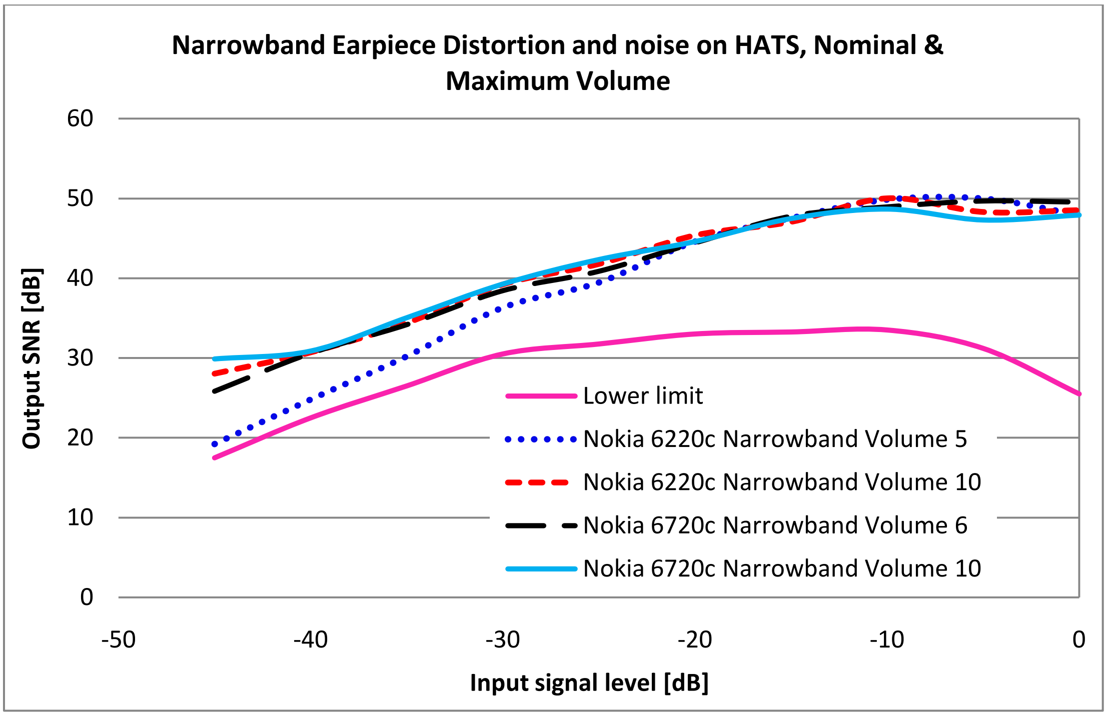


Figure 30: Measured earpiece distortions on narrowband on nominal and maximum volumes. Narrowband bit rate was 12.20 kbit/s and wideband bit rate was 12.65 kbit/s.

The difference between narrowband and wideband measurement result on the Nokia 6220c is emphasized by the volume level. The lower the volume, the more distortion there is in lower signal levels.

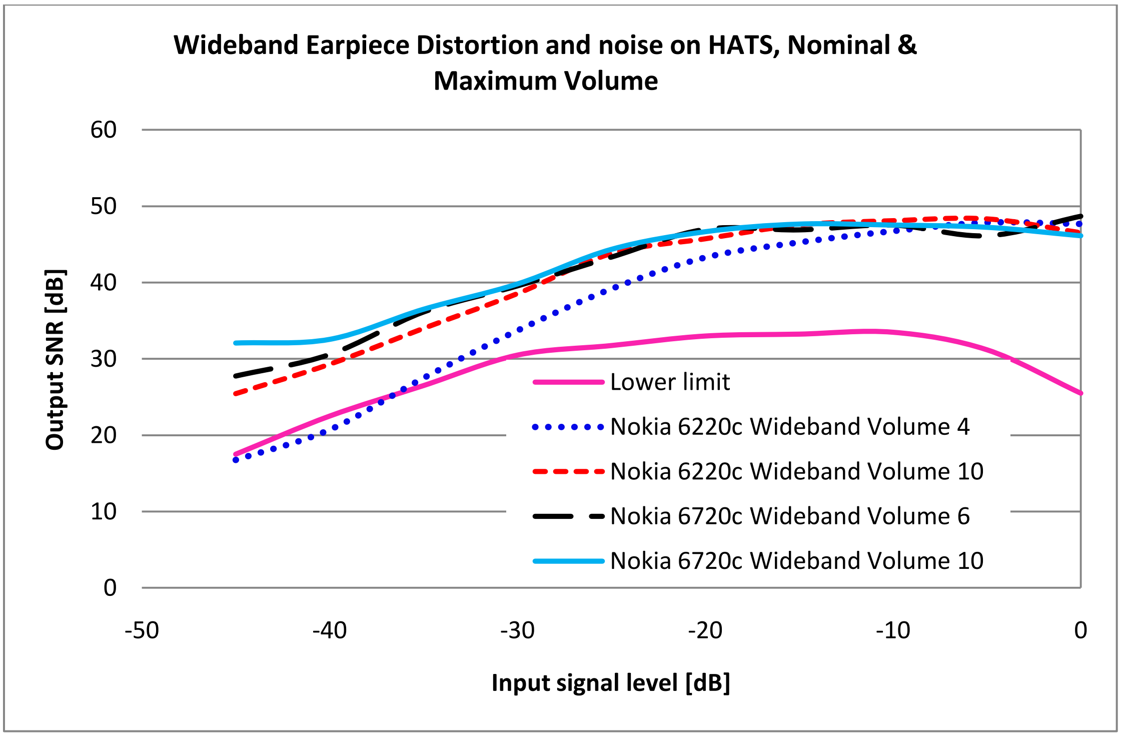


Figure 31: Measured earpiece distortions on wideband using nominal and maximum volumes. Narrowband bit rate was 12.20 kbit/s and wideband bit rate was 12.65 kbit/s.

6.5 Discussions about the objective measurements

The objective measurements revealed expected differences between the small and large speaker in capabilities of producing sound. The main reason for the differences in results is the size of the earpiece membrane, which is discussed in the loudspeaker theory Section 5.2.1 and seen in the free field measurement without being integrated to the phone results in Section 6.2. After these it could be expected that there will be differences in the results when the speakers are integrated to the phone and measured on HATS.

Measurements on HATS illustrate the meaning of acoustics. If the measurements without the phone audio processing are compared to 3GPP measurements where the phone equalization and other audio enhancements are online, the results revealed that audio processing can help to fix the problem by making the frequency response flatter. The reason the frequency response is aimed to be flat is simply to avoid emphasizing or diminishing certain frequencies, which could harm the understanding of hearing the speech.

The incapability of reproducing low frequencies at a satisfying level for wideband speech on a small earpiece can be seen from all measurements done in this chapter. The distortion levels at different input voltage levels in Figure 26 revealed the unlinearity of distortion on the small earpiece under frequencies of 450 Hz when the large earpiece has fairly predictable levels until 100 Hz. In practice, the small earpiece may produce audible distortion at low frequencies when the user sets the phone volume level to maximum. Naturally, this limits the maximum output level that a designer can allow to come out from the small earpiece without distortion. The other option is to damp the low frequencies and allow the higher frequencies to sound louder. This option is recommended only for narrowband speech and in the Nokia 6220c, the narrowband speech is tuned to produce more higher frequencies (500 - 3000 Hz).

The differences of the two phone earpiece integration is shown on a theoretical level and through objective measurements. The larger earpiece integration is 10 dB louder at 100 Hz and the fundamental frequency of male speech is around 100 Hz. After these facts it can be said that the end-user should hear audible differences between the realizations. The next chapter tries to find the answer to this and it is interesting to see the subjective test results.

7. SUBJECTIVE TEST ON TWO EARPIECE INTEGRATIONS

The background of arranging the listening test is presented and the phases of processing the subjective test files are described. In the end the listening test results are shown to support the objective results.

7.1 Overview of subjective testing

Subjective testing is used in situations where there is no well-proven objective measure of audio quality. The objective data can be used for speaker comparison, but the objective measurements cannot be used in determining every audible characteristic of the speaker. The problem is that human ears with brain analyzing do not process the sound as microphones or measuring instruments do. When consumer buys a mobile phone and listen to the speaker audio quality, the objective measurement data is not available and this way the subjective experience becomes more important.

Subjective testing is a time consuming process, because a subject grades the performance of many samples. Usually, the collected result data is quite sparse and some variation occurs in identical samples due to the subject's personal opinion. However, proper test planning and constant testing conditions can decrease the variation in the test results.

Performing a subjective test in an efficient way is a strict process. In [15] the following procedure is suggested:

- Definition of what is to be tested and null hypothesis
- Selection of test paradigm
- Creation of test material
- Definition of sample population
- Selection of listeners
- Familiarization and/or training of subjects
- Running the test
- Analysis and reporting of results

7.1.1 Test type and listener selection

It is important to know that the selected test type affects the time consumption for testing. There are several test methods to use for subjective testing [15]:

- *Single stimulus*, the mean opinion score (MOS) test belongs to absolute category rating. Only one sample at a time is played and evaluated. The benefits are fast speed and absolute rating.
- *Paired comparison*, A/B tests are for comparing relative quality of two samples A and B. A few different possible comparison methods are:
    - *A or B*, two possible options which is better.

  - *A or B scale*, both samples are rated with the same scale.
  - *A or B scale with fixed reference* is a test where one of the test samples is the known reference and the other is a degraded from the reference.
  - *A or B scale with hidden reference* is same as the previous but the reference is not known, which enables either samples to be better.

- A, B or X test is made up of three samples. The idea is to choose, which one of two samples, A or B is closer to the quality of X.
- *A, B or C* test is a triple stimulus comparison with a hidden reference. Two of three samples are similar and one is the reference. The listener decides, which sample of the other two samples is different to the reference and evaluates the sample.
- *Rank order* has several stimuli to compare. The relative order of the stimuli is rated. Rapid ranking is a method of quality. The downside of it is that the perceptual distance is unknown between the stimuli.

When the samples for the listening test were processed, it was time to decide the listening test type. Because the differences between the small and large speaker are rather small, the sensitive test type was needed. That's why the most convenient test type for our purpose was A or B scale with a hidden reference. This method is quite slow, but by choosing suitable amount of samples, the test duration was about 30 min long.

The DaGuru listening test software has a few different choices for the mentioned (A/B with hidden reference) test method and comparative mean opinion score 3 (CMOS3) was selected. CMOS3 has a scale from -3 to 3, which was good enough for the purpose. Choosing -3 means a lot worse and 3 a lot better than the first sample.

An important matter when setting up a listening test is the question of how many subjects have to be recruited. Selecting the sample population type affects to the decision about the size of the population as well as the reliability of the results. In [19] listeners are divided into three groups:

- Naïve
  - Subjects have not been selected for any discrimination or rating ability
  - Subjects belong to the general public
  - Discrimination and reliability skills are unknown
  - In order to obtain low error variance, 24-32 listeners are required
- Experienced
  - Subjects have experience in listening to a particular type of sound or product
  - Experience does not promise reliability and repeatability
- Expert
  - Tested subjects with normal hearing, good discrimination skill and reliability
  - Subject may be over sensitive to aberrations in samples
  - 10 subjects is enough for low error variance

7.2 Creation of test material

Before creating the listening test material the listening method had to be chosen. Several different testing methods were discussed with the instructor of this thesis and Nokia colleagues, resulting in three final options:

1) *A real phone*, listener hears a sample from a real phone in call, which is changed to the other one after one sample is heard.

+ Real speaker implementation and call, speaker differences are possibly heard easier in a noisy environment
+ Most accurate method as all interferences and problems of recording environment are avoided
- The look of the phone affects the results
- Hard and laborious usage in the test, phones have to be switched many times

2) *Rapid model*, large and small speakers are integrated to identical rapid models, which are changed after the sample is heard. Signal processing is done on the samples before the test. Finally, samples are played from a laptop and fed to the speakers.

+ Real small and large speakers in use
+ The phones' look is identical and does not affect the evaluation
- Requires pre-work and signal processing before the files are ready for listening
- Hard and laborious usage in the test, phones have to be switched many times

3) *Headphones*, samples are recorded with HATS and processed before the test. Samples are played from a laptop with headphones for the listeners.

+ Easy for listener
+ Listening test software can be used
+ No phone switching and effect on results by look
- Requires lots of work before the samples are ready for listening
- No real implementation
- Stages in signal processing diminishes large speaker advantages against small one
- Recording environment and sample processing adds interference to samples

Despite the heavy processing of samples the headphones were selected to be the listening method for the samples. The ease of listening and answering with listening test software was a big criteria to end up selecting the headphones. There was a plan to do a listening test also in background noise and test intelligibility differences. Usage of headphones was not feasible as the headphones damp the background noise.

Because the listening test was decided to proceed with headphones instead of using real phones or rapid models, the sound files had to be processed to sound like a real phone call in the listeners ear. Pre-processing the sound files was a laborious process, which is described in the nine-step block diagram in Figure 32.

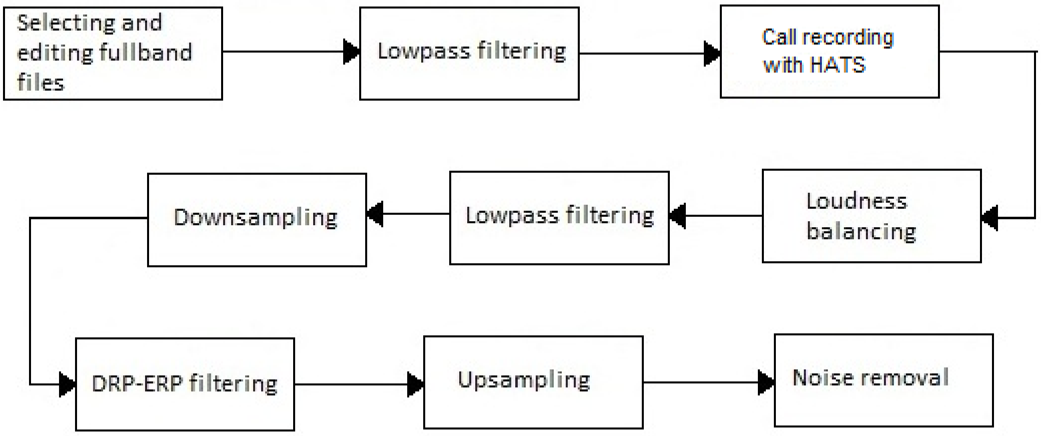


Figure 32: Block diagram of all stages of creating listening test samples. More detailed description of every stage is presented in the following sections.

7.2.1 Full band files and selecting test samples

The listening test language was decided to select to be Finnish because finding 30 native English speakers for the listening test would have been a harder task to complete. Luckily, the material for arranging the listening test was recorded earlier and selecting suitable sentences was a fairly easy task.

The sound files used in this test were recorded in Tampere Nokia premises in an anechoic chamber. There were 4 female and 4 male speakers available to choose from reading the same texts. From these 8 readers, 2 male and 2 female speakers were chosen for the listening test. Speakers were chosen for this test in a way that their pronouncing was clear. The second criterion was that from both gender one speaker had more low frequencies and the other had more higher frequencies. This was decided by examining the spectrum of each speaker's voice averaged over all sentences found from [19].

The speech samples available were several sentences long so the suitable two sentences with duration about 10 seconds were selected. There were several different themes available and the purpose was to select different one for all 6 sentences. This worked out very well and only tar was the subject in two sentences. The idea was to keep listeners awake by altering the themes. The sound sample editing process was done on Adobe Audition 1.5.

7.2.2 Lowpass filtering to 7.8 kHz

The fullband files had to be lowpass filtered for the radio communication tester to avoid any possible errors to the sound signals. The selected samples were filtered using a lowpass finite length impulse response (FIR) equiripple filter created on Matlab. The Matlab code and more specific information about the filter is found from Appendix D:. The filter magnitude response is shown below.

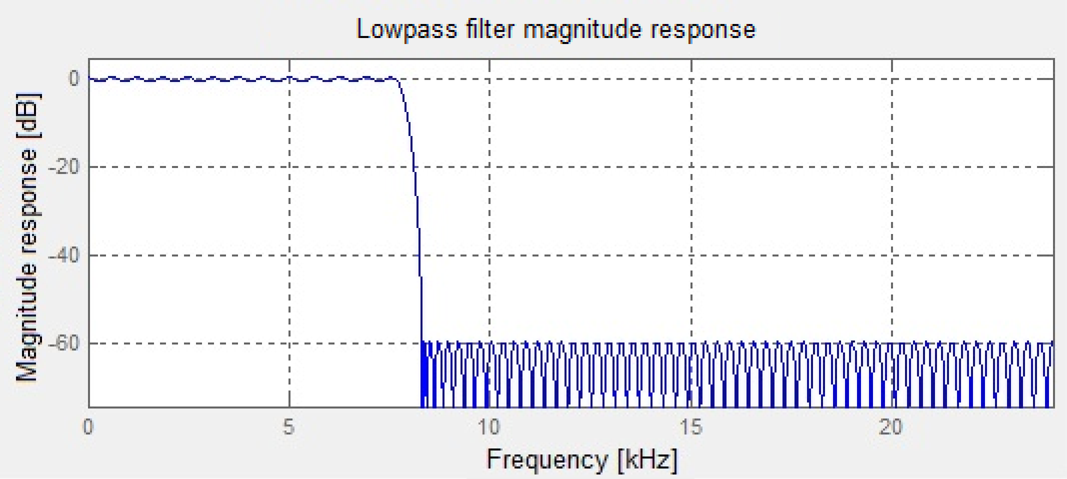


Figure 33: The 7.8kHz lowpass filter magnitude response used to the full band samples. Sampling rate was 48000Hz.

7.2.3 Call recording with HATS

The filtered files were played from a laptop and fed to a Rohde & Schwarz radio communicator tester CMU. The laptop output signal level was adjusted according to the CMU data sheet [37] to full-range input level in low sensitivity mode. The peak voltage level is 1.4V and the laptop soundcard output RMS voltage level was supposed to be -19.14 dBFS to prevent clipping of the speech signal. The required voltage level is calculated from the following equation

$$G_{dB} = 20\log\left(\frac{x}{V_{ref}}\right) \tag{32}$$

$$-19{,}14dB = 20log\left(\frac{x}{1{,}4V}\right) \tag{33}$$

$$x = 1{,}4V * 10^{\left(\frac{-19{,}14}{20}\right)} \approx 155mV \tag{34}$$

where G_{dB} is the desired CMU input gain level in decibels, V_{ref} is the CMU maximum input voltage peak level and x is the CMU input voltage level for the desired decibel value -19.14 dBFS and also the soundcard output voltage level. The result 155 mV was measured from soundcard output when a -19 dBFS multitone was played by Adobe Audition. The audio signal level in the phone's DSP was traced with a tracing device (called Musti) to make sure the signal is not clipped before phone the earpiece. After tracing showed that everything is in order, the phones were positioned to HATS and a multitone signal was played from the laptop and volume levels were adjusted to be at a suitable level for the actual recording process.

The CMU was used to make the call to the phone in a GSM 900 and WCDMA network. Each phone was positioned to HATS according to the designer’s directions. The received sound sample was recorded with HATS and amplified 20 dB on the B&K Nexus amplifier. Finally, the samples were recorded on laptop using Adobe Audition. A more detailed measurement setup is shown in Figure 34 and the equipment is described below.

Table 8: Equipment used to record speech from phone earpiece placed on HATS.

Instrument	Type	Comment
Radio communication tester	Rohde & Schwarz CMU 200	GSM & WCDMA network used
Head And Torso Simulator	B&K	Includes artificial ear with microphone
microphone amplifier	B&K Nexus	HATS mic amplifier
Sound card	VX pocket 440	PCMCIA card with IBM thinkpad T41
Phone signal tracker	Musti	Speech signal level traced on phone DSP

Phone volumes were selected to be 6/10 and 10/10, because volume 6/10 is the level that user would normally use in daily life and 10/10 is for the noisy environment. The maximum level 10/10 was selected to find out if there is audible distortion in the recordings with the small earpiece. Also, the speech signal level was higher and the background noise was about the same, i.e. SNR was better than 6/10 level.

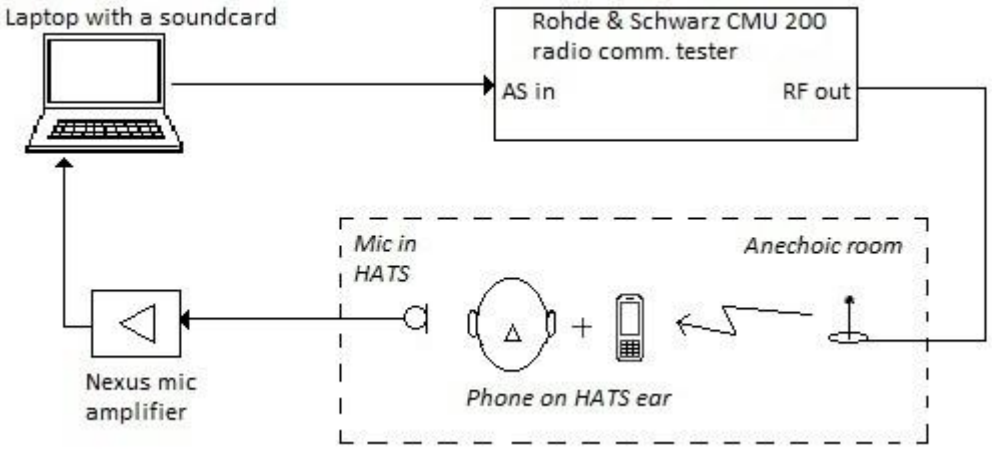


Figure 34: Measurement setup for live call recording with HATS. AS = Analog Signal, RF = Radio Frequency.

7.2.4 Loudness balancing

When the samples were recorded with HATS, both the nominal and maximum volume levels were used in the phones. It is said in [1] that usually the louder the sound is the better it sounds to the listener. By aligning the sound level the loudness difference affect is minimized and listeners can concentrate on evaluating correct affairs.

The balancing was done using the loudness batch tool version v1.4 and the following parameters were given to the program: Input filename, align all samples to 27 Moore average sones, align all samples to 27 Moore dynamic sones, align all samples to within 15% of the target sones. These parameters resulted in files having a peak amplitude of -6±1 dBFS and an average RMS power of -31±1 dBFS.

7.2.5 Lowpass filtering, wideband 7.8 kHz, narrowband 5.7 kHz

There was some background noise from cables and measurement equipment added to the recorded signal during the call recording with HATS. Also, the phone audio hardware causes noise to the output signal. Because of the loudness balancing the noise outside the speech signal was amplified and it had to be removed with a FIR equiripple lowpass filter. Both narrowband and wideband signals had their own lowpass filters. The cutoff frequency for narrowband samples was 5.7kHz and for wideband 7.8kHz.

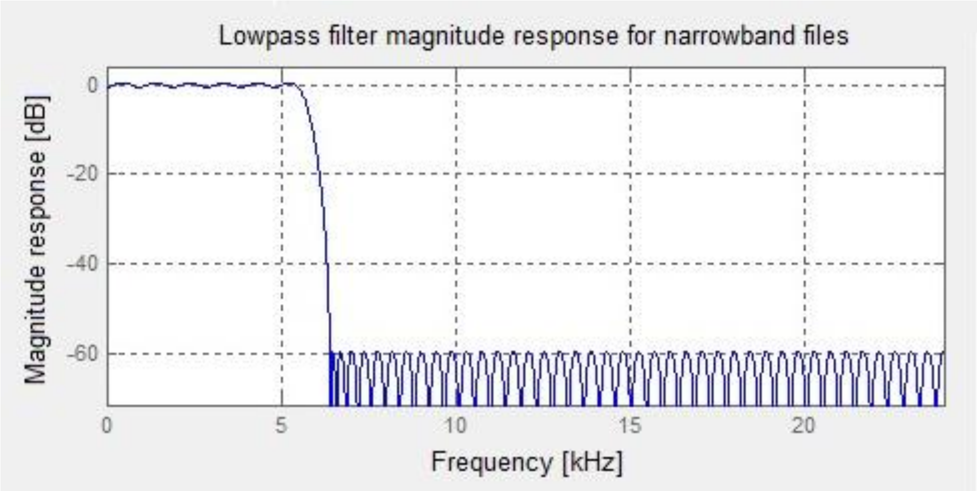


Figure 35: Lowpass filter for narrowband samples used to decrease noise outside the speech signal. Sampling frequency was 48000 Hz.

7.2.6 Downsampling from 48 kHz to 24 kHz

This operation had to be done because the filter in the next step in Section 7.2.7 had problems following the measured frequency response of the HATS ear and headphones. The options were to select either a high sampling rate 48 kHz with the loss of filter accurate on low frequencies or lower the sampling rate to 24 kHz and reproduce the low frequencies fairly near to original frequency response. In this case, the low frequencies were more important because the main differences between the speakers are under 1 kHz. Moreover, the sound quality was very good at 24 kHz sampling rate, because it is more than 16 kHz minimum sampling rate that wideband requires. The tool for downsampling was the *ReSampAudio* batch tool, which required only to input a file name, new sampling rate and output file name for parameters to complete the process.

7.2.7 HATS ear canal (DRP-ERP) filter

An important phase in processing the recordings was to eliminate the effect of the HATS ear and the headphones from the recorded sound files. For creating the filter, the Sennheiser HD256 headphones were placed to the HATS ears and the frequency response of those together was measured as in Figure 36. The measurement equipment was the same as in Table 3 in full audio path measurements to phone measurements on HATS.

The measured frequency response was inverted to compensate the effect of the HATS's ear and headphones. To create a filter of the measured data, the data was read to Matlab for further processing. The equalizer was created from the read data by a modified Yule-Walker algorithm, which is an add-on to Matlab. The algorithm could not perform at the desired 48 kHz sampling frequency as at 24 kHz and the downsampling had to be done as mentioned in Section 7.2.6. The parameters used for the algorithm are presented in Table 9 and the result is in Figure 37.

Table 9: Parameters in Yule-Walker algorithm used for filter designing to headphones on HATS ear.

Sampling frequency [Hz]	24000	Other options	
Lower do not care limit [Hz]	100	0 dB loudness correcting	on
Upper do not care limit [Hz]	8000	Uniform weighting	on
Filter order	12		

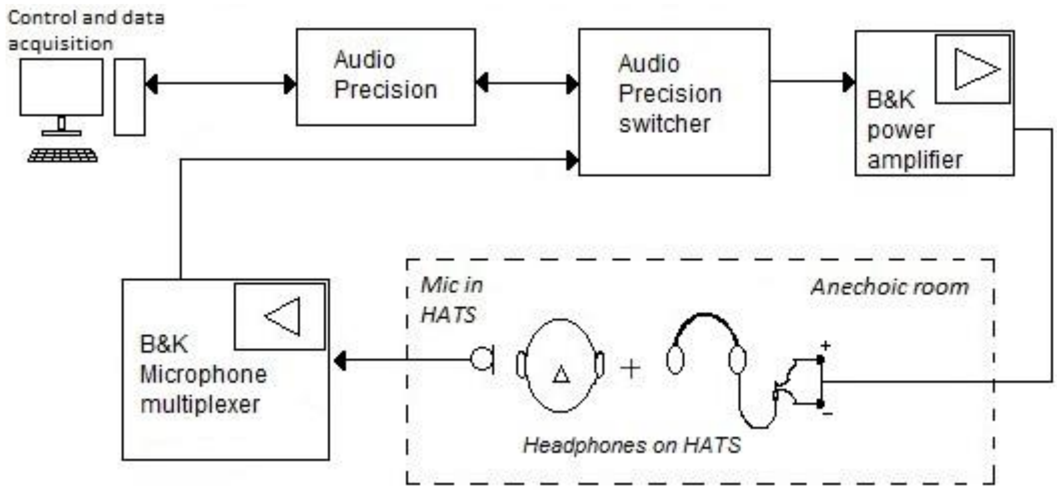


Figure 36: Measurement setup for headphones on HATS.

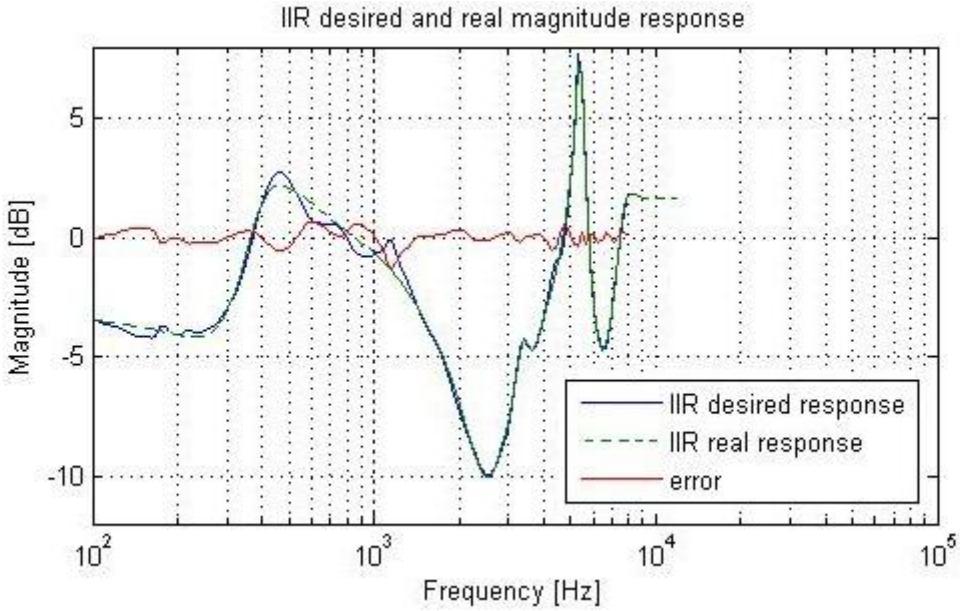


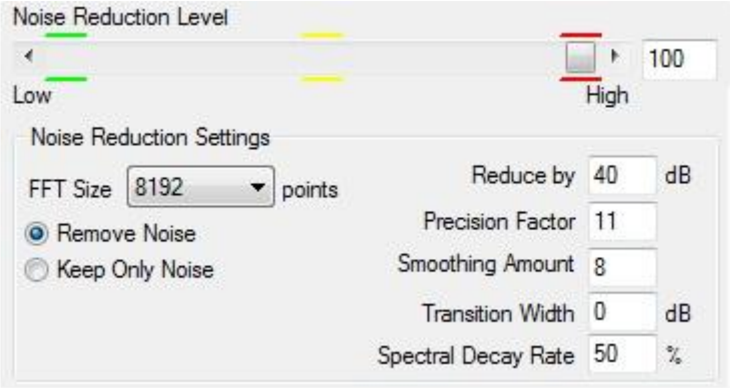
Figure 37: Filter magnitude response made from measured Sennheiser HD256 + HATS drum and ear reference point data.

7.2.8 Upsampling from 24 kHz to 48 kHz

It was nice to notice after all these stages that DaGuru listening test software was unable to play the 24 kHz sample rate files. Therefore, the upsampling operation was done purely because of the listening test software. On the other hand, the software was good and easy to use so the upsampling was a natural choice instead of finding another software tool. The same tool used for downsampling (*ReSampAudio*) was used for upsampling the files from 24 kHz to 48 kHz.

7.2.9 Noise removal

After upsampling, the sound samples were fairly noisy especially around 5 kHz because the HATS *ERP-DRP* filter emphasized those frequencies. In narrowband, that peak was outside the speech band and it was very disturbing to listen to. The noise was reduced by a noise removal tool found from Adobe Audition 1.5.



Noise Reduction Level

Low — High: 100

Noise Reduction Settings

FFT Size 8192 points

● Remove Noise

○ Keep Only Noise

Reduce by 40 dB

Precision Factor 11

Smoothing Amount 8

Transition Width 0 dB

Spectral Decay Rate 50 %

Figure 38: Adobe Audition noise reduction parameters used to the listening test samples.

7.2.10 Discussion of the listening test files

After all stages described in Sections 7.2.1-7.2.9, the samples were ready for the listening test. The two figures below are shown to demonstrate the magnitude response of male and female speakers on both phones after all recording processing was done as presented in Section 7.2. Figure 39 presents the 10 second long averaged wideband male and female samples from both phones.

The differences between the phones are shown in Figure 40 by subtracting the Nokia 6220c results from the Nokia 6720c results (large speaker - small speaker). Also, the subtraction between the male and female samples are shown to demonstrate the different magnitude responses between the gender of the speakers. By looking at Figure 40, it can be seen that the Nokia 6220c emphasizes the frequencies between 200-500 Hz and the Nokia 6720c 1.5-4 kHz and especially frequencies below 150 Hz. The greatest difference between the samples from the different phones is about 10 dB and it should be heard easily by the listeners. Therefore, it is interesting to see the subjective results and hear the comments about the samples.

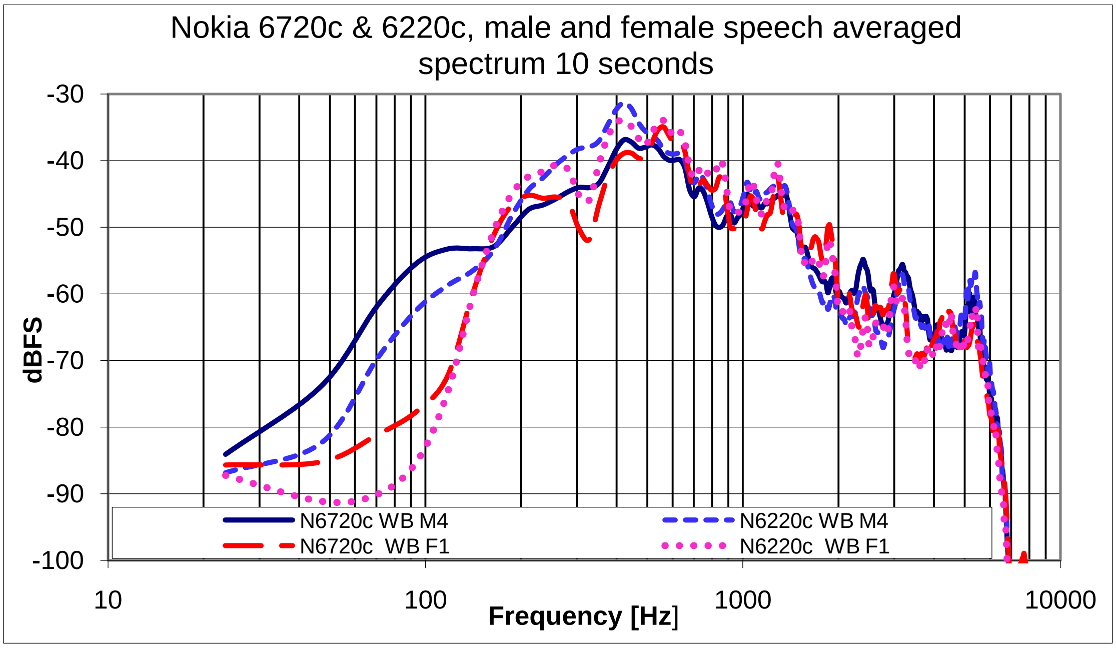


Figure 39: *10 seconds long averaged wideband male and female samples from both phones after all recording process was done ready for the listening test. Wideband bit rate was 12.65 kbit/s.*

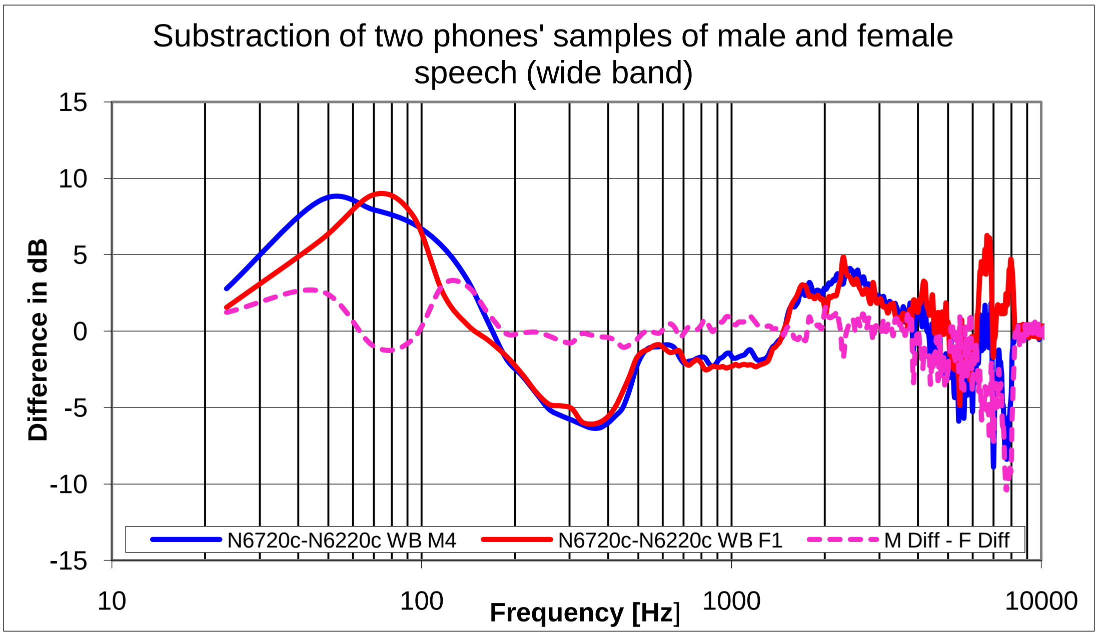


Figure 40: *Subtraction of Nokia 6720c and 6220c speech samples on wideband. The results of the same gender is shown with a solid line and male-female subtraction (M diff - F Diff) result is marked with a dash line.*

7.3 Listening test set-up

The listening test was held in Oulu Nokia (Elektroniikkatie) premises, in the listening room in F-wing during 1.-20.10.2009. The dimensions of the listening room are 440 x 794 x 242 (w x l x h) and so the volume is approximately 85 m^3. The listening position was set to the middle against of long wall, where the laptop, mouse and headphones were located.

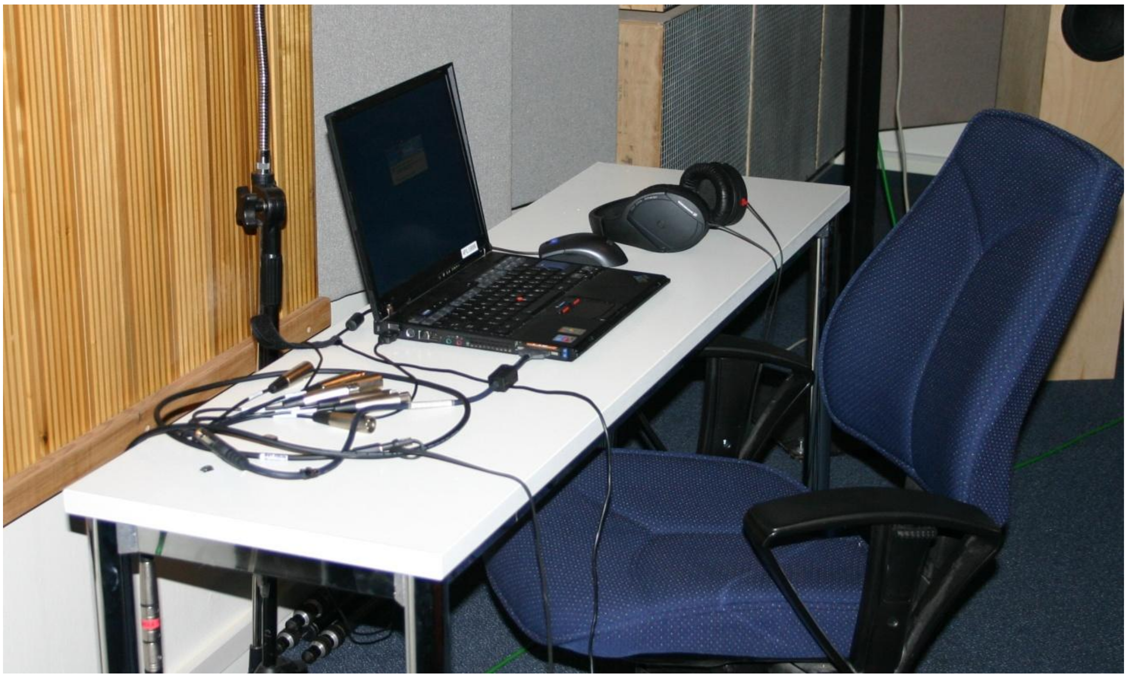


Figure 41: Listening test position in the listening room.

Since the discrimination skills of the listeners were not known, 2 additional sample pairs (1 narrowband & 1 wideband) with identical samples were included to the listening test. There were also 4 sample pairs recorded on the same phone, but on different volume levels 6/10 and 10/10. These sample pairs recorded on the same phone volume levels were matched with the loudness tool described in Section 7.2.4. By looking at the evaluation results of these identical and same phone samples, the skills of individual listeners were somehow concluded.

7.3.1 Briefing for the listener

In order to get listeners to focus on the desired parameters, short briefing has to be held before the test. Also, the technical issues were discussed and clarified, like how to use the test software. If these preparations are skipped, the users may be uncertain how to interact with the software and the test results may vary a lot depending on the listener. To avoid any affects on the results just because of lack of information, the following items were discussed before the test.

- Test method
- Content of the test
- Parameter what listener should evaluate
- Possible imperfections that should not affect the evaluations
- What is prohibited
- Staff location, who to ask if any problems occur

The headphones orientation and position were instructed to the listeners before the test. Also, the user interface was demonstrated to avoid causing confusion to the listeners.

7.3.2 User interface

Because all the listeners and samples were Finnish the user interface language was set to Finnish. The listening test software was DaGuru and it was run on a normal laptop with Windows XP operating system.

Using the software was made easy for the listener. First the name and age was given to the software and after that 8 practicing sample pairs were played. After practicing, the listener had to listen and rate 50 sample pairs, which took about 30 min. The user had to listen to a sample pair and grade the recent sample compared to the previous one. When the sample pair was evaluated, the next pair was played automatically. If a user wanted to listen to the sample pair again, they were instructed not to use more than 2 repetitions at save time. The number of evaluated and upcoming samples were on the screen all the time as can be seen from Figure 42.

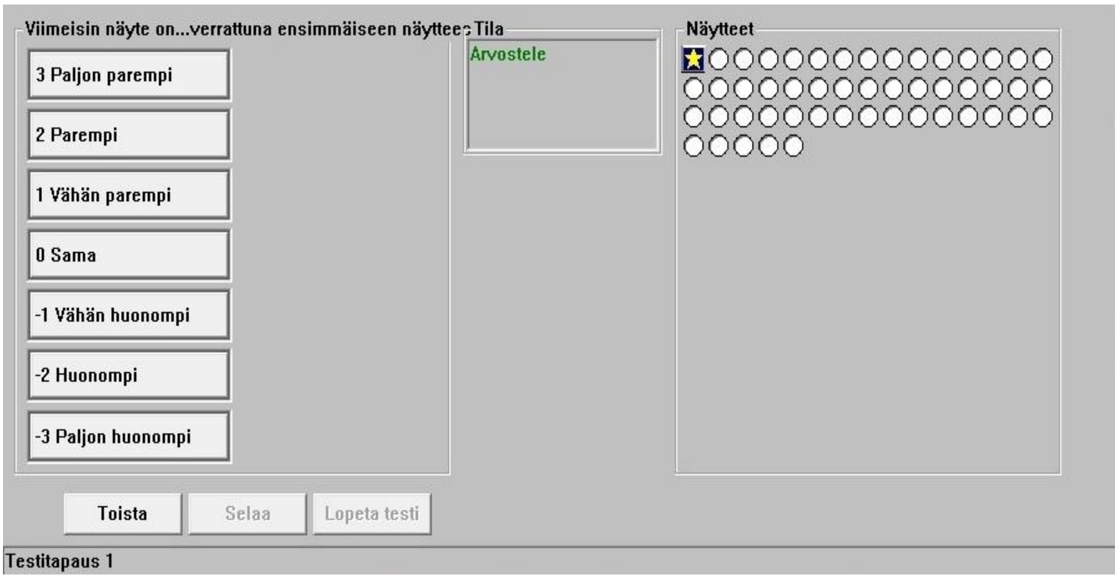


Figure 42: User interface in the listening test. Two samples have been played and the listeners have to choose how good the latest sample is compared to the previous one. Users can repeat the sample pair by pressing the "Toista" button.

7.4 Results of the subjective test

After all preparations were done, the subjects took part in the listening test and finally, the analyzing was proceeded. Altogether 33 subjects took part in the test, from which 17 were male and 16 female. The analyzing process and results of the listening test are presented in this section.

7.4.1 How the data is analyzed

The analysis of the test results was done on Microsoft excel. DaGuru has an excel add-on, which reads the listening test data for analyzing. The add-on assorts the answers by using the first sample as a reference sample and the other as a test sample, which is evaluated. The Nokia 6220c was selected to be the reference (just in the analyzing phase) and the grading was executed in the following way: If a listener has given grade 3 to a test sample (Nokia 6720c), the answer is Nokia 6720c grade 3. If the reference and test sample are the other way around, the inverse grading result was given to test sample as in Table 10.

Table 10: Simplified table of results analyzing process.

Listener			Analyzer	
First sample	Second sample	Grade given to second sample	Result	Grade
Nokia 6220c	Nokia 6720c	2	Nokia 6720c	2
Nokia 6720c	Nokia 6220c	-3	Nokia 6720c	3

After the listening test results were arranged as shown in Table 10 the mean opinion score, standard deviation and 95% confidence interval were calculated with Excel. The equations were the following [20]:

- Mean:

$$\bar{x} = \frac{1}{n}\sum_{i=1}^{n} x_i \tag{35}$$

- Standard deviation:

$$\sigma = \sqrt{\frac{1}{n}\sum_{i=1}^{n}(x_i - \bar{x})^2} \tag{36}$$

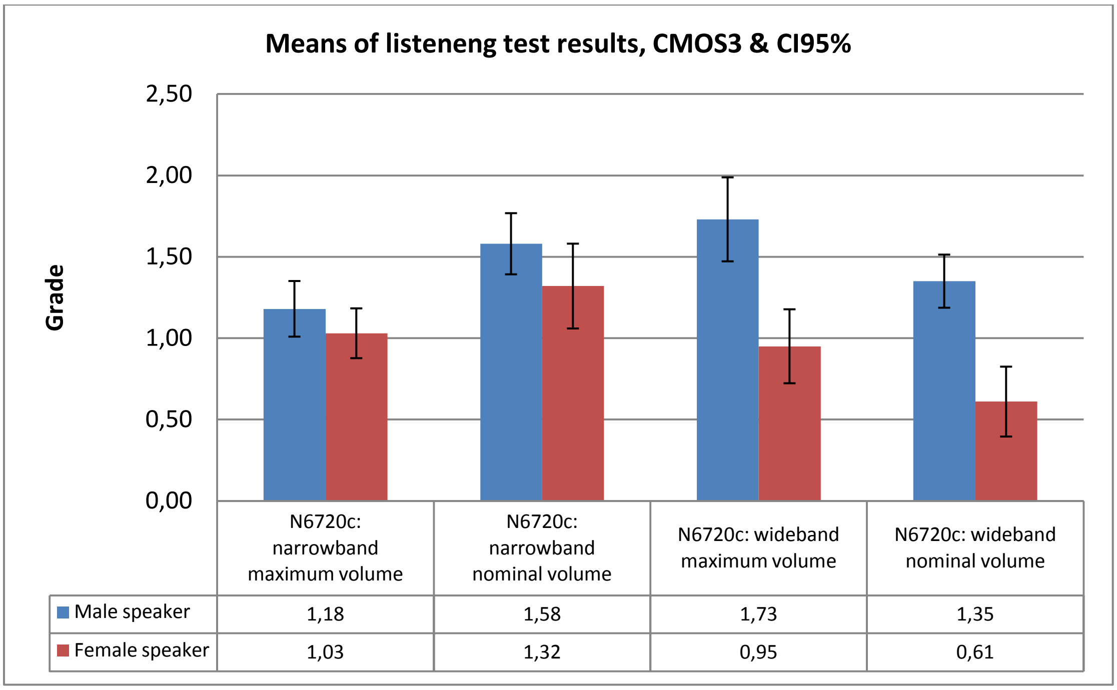
- 95% confidence interval:

$$C_{95\%} = \bar{x} \pm 1.96(\frac{\sigma}{\sqrt{n}}) \tag{37}$$

where x is vector with all the results, $\bar{x}$ is the mean value of vector x, n is the number of given grades, σ is standard deviation and $C_{95\%}$ is the 95% confidence interval. The purpose of the confidence interval is to indicate the reliability of the results by giving a region where the true value of the evaluation result is in probability of the given percentage value. A 95% confidence interval is often used and if the two confidence intervals do not overlap in the two evaluated items, the values of the items are said to be significantly different.

7.4.2 Analysis of the results

After all the 33 listeners had taken part to the listening test, it was time to analyze the results. There was 17 male and 16 female listeners and the target was 15 male + 15 female listeners. Because there were 2 sample pairs with identical samples, it was easy to select the 2 male and 1 female speakers out from the 33 listeners, who heard the both samples worse (-2) or better (2).



	N6720c: narrowband maximum volume	N6720c: narrowband nominal volume	N6720c: wideband maximum volume	N6720c: wideband nominal volume
🟦 Male speaker	1,18	1,58	1,73	1,35
🟥 Female speaker	1,03	1,32	0,95	0,61

Figure 43: Means and 95% confidence intervals of all the scores given for the Nokia 6720c compared to the Nokia 6220c. Nominal volume means volume level 6/10 and maximum 10/10. The *Nokia 6720c got better scores in every case evaluated.*

When looking at Figure 43 the it should be kept in mind that N6720c has grades from 0.61 to 1.73 and N6220c has thereby inverse grades. It can be noted that male speakers on wideband benefit significantly more from the large speaker. On the other hand, on narrowband the benefit from the large speaker is about the same for both male and female speakers.

In addition to naive and experienced listeners, 7 people from Nokia Oulu (Teknologiakylä) electro mechanic audio team were invited to the test. The reason for inviting those 7 people were partly because the listening test arrangements had to be tested. One other important matter is to obtain information how close the designers evaluate the test samples compared to the results of other listeners. The results of all listeners are presented in Figure 44.

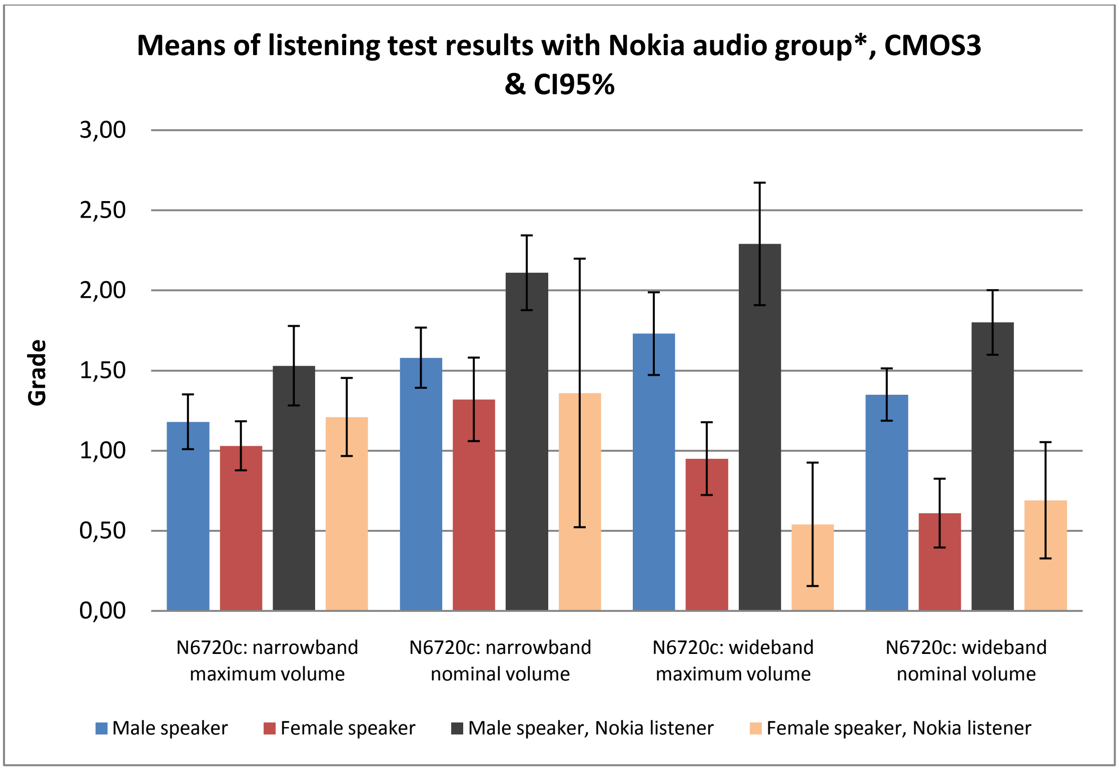


*Figure 44: Means and 95% confidence intervals of all the scores given for the Nokia 6720c compared to the Nokia 6220c. Nominal volume means volume level 6/10 and maximum 10/10.*7 people from Oulu Nokia (teknologiakylä) electro mechanic audio team took part in the test.*

7.5 Discussions about the subjective test

Using the headphones instead of real phones probably affected the test results. Nokia 6220c had audible background noise especially with female narrowband speech, which may have affected that the Nokia 6720c female speaker narrowband quality was evaluated better than wideband quality. The standard deviation became fairly large on narrowband nominal volume among the Nokia audio team listeners, because there was audible noise in some of the nominal volume listening test files even after noise reduction. The listeners were instructed not to take the noise into account, but still it has probably affected the grading. The small population size (7) also has an effect on the audio team confidence interval.

The processing of the files was an easy task excluding the small noise problem with the narrowband files. The listening experience could have been better if the loudness correcting would have been done as the last process before the listening test. A few listeners said that a couple of files sounded to be at a different volume level, which could have affected the results.

The listeners said some comments about the darkness of the female sound after the test. In practice, all the lowest frequencies were produced by the Nokia 6720c. There is a possibility that low frequencies were emphasized too much for a female speaker and caused the feeling of dark sound. That may have caused the result that female speaker audio quality on both phones were almost equally graded, i.e. near zero on wideband.

8. CONCLUSIONS

The objective of this thesis was to compare the earpiece integrations in a mobile phone subjectively. This work contains theory about earpiece integration in mobile phone with the analysis of objective and subjective measurements. The results of the subjective test indicate that end-users can distinguish between the different earpiece integrations.

In Chapter 6 the objective measurements were performed and some expectations of subjective test results were done. The measurements on HATS showed considerable differences on earpiece integrations. Low frequency reproduction on smaller earpiece integration was poor due to the earpiece physical limitations discussed in Chapter 5.

In the light of theory and objective measurements, the results of the subjective test were not a surprise. Male speaker benefit clearly because the low frequencies were produced much better than the small speaker implementation. Listeners experienced the female speakers sound as being too dark, but this can be due to recording, processing and using the headphones for listening to the test files. It was interesting to try how the audio designers' grades differentiate from normal user listening test results. The Nokia electro mechanic audio team from Oulu participated in the test with 7 people and graded the samples with the same trend as the other listeners.

As discussed in Chapter 7, concerning different listening methods, the headphones were selected to be the most convenient to use for the listener. However, using another listening method could produce even better grades for larger speaker integration. It would have been useful to try the listening test with the rapid model under silent and noisy conditions. The intelligibility test could have been added to the noisy conditions to see if end-users rate the earpiece integrations the same way as in silent environment.

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10. APPENDICES

APPENDIX A: Large speaker used in Nokia 6720c mobile phone

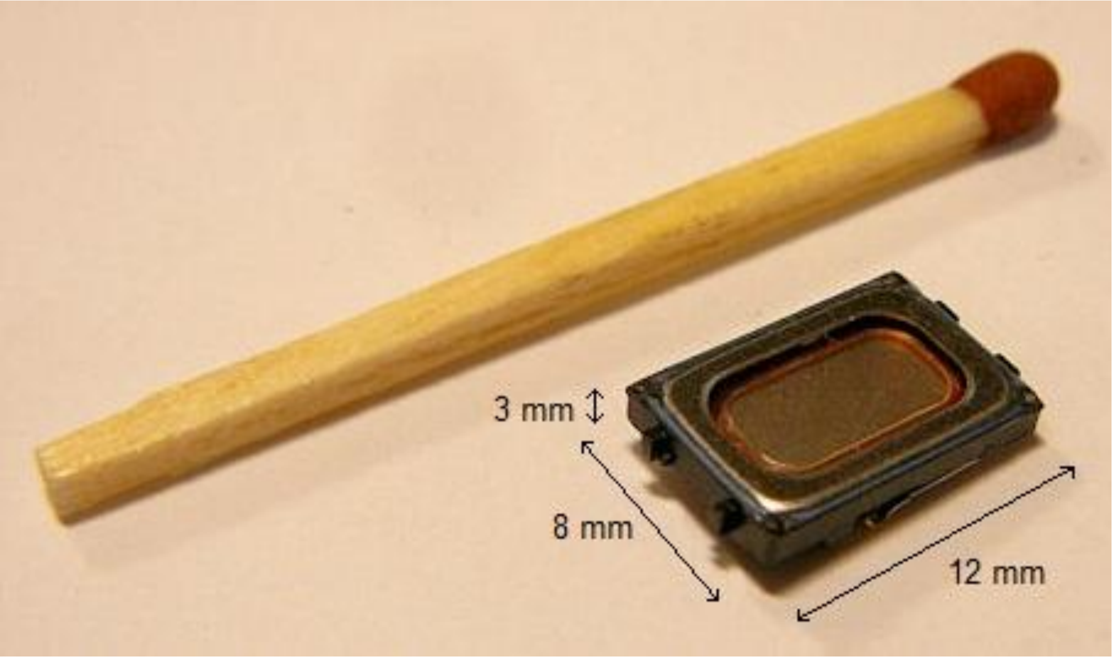


Figure 45: Large speaker dimensions used in Nokia mobile phone in listening test. The dimensions are taken from [16].

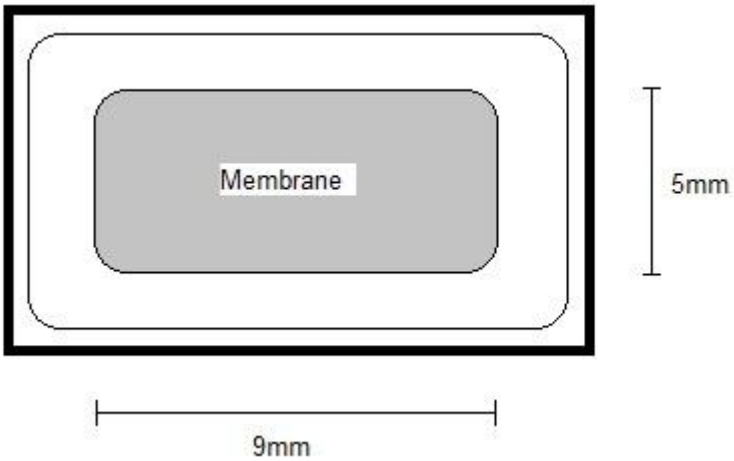


Figure 46: Measures of the large speaker membrane used in Nokia mobile phone in listening test.

APPENDIX B: Small speaker used in Nokia 6220c mobile phone

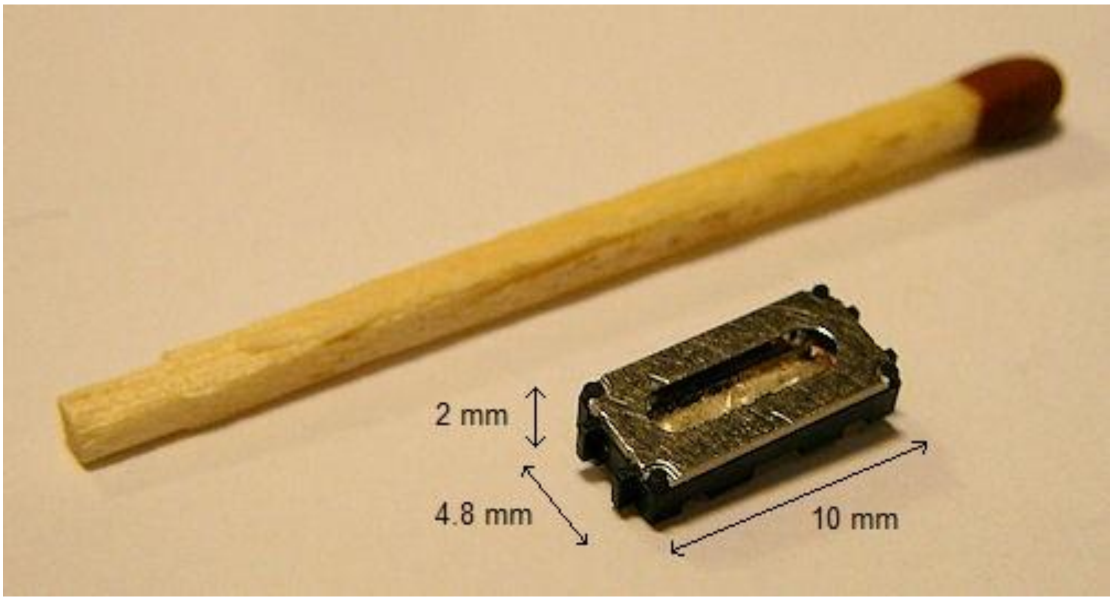


Figure 47: Small speaker dimensions used in Nokia mobile phone in listening test. Dimensions are taken from [17].

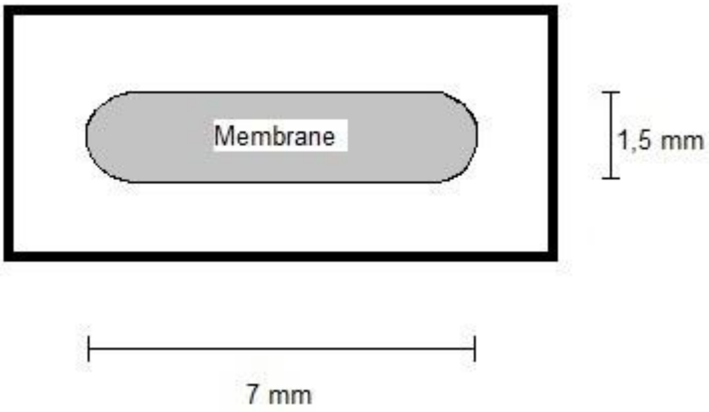


Figure 48: Membrane measures of small speaker used in Nokia mobile phone in listening test.

APPENDIX C: Listening test instructions (in Finnish)

Kuuntelukoeohje

Tässä kuuntelukokeessa kuunnellaan suomenkielisiä puhelinääninäytteitä. Näytteet soitetaan pareittain, joista arvioidaan jälkimmäisenä kuultua näytettä verrattuna ensimmäiseen asteikolla -3 (Paljon huonompi)…3 (Paljon parempi). Jos jälkimmäinen on mielestäsi huonompi kuin ensimmäinen, valitse -3…-1. Jälkimmäisen ollessa parempi kuin ensimmäinen, valitse 1...3. Jos et kuule eroa näytteiden välillä, valitse 0. Arvioinnin perusteena on, kumpi näytteistä kuulostaa mielestäsi paremmalta.

Äänenvoimakkuus säädetään testin alussa kuuntelijalle sopivaksi, eikä sitä tarvitse muuttaa testin aikana. Jos haluat kuulla näyteparin uudelleen, voit toistaa sen enimmillään 2 kertaa, jottei testi mene liian pitkäksi.

Kun kuuntelukoe alkaa, ohjelma kysyy nimeäsi ja ikääsi. Voit kirjoittaa sen muodossa NimiS 15V, jossa S on sukunimen ensimmäinen kirjain. Aluksi kuunnellaan 8 harjoitusnäyteparia, jonka jälkeen varsinainen testi alkaa. Tämän jälkeen kuunnellaan 25 paria nykyisin puhelimissa kuultavaa puhelinääntä, joiden jälkeen tulee 25 paria tulevasta laajakaistaisesta puhelinäänestä.

Joissakin näytteissä voi olla häiriöitä (epäjatkuvuus, kohina), joita ei tarvitse huomioida näytettä arvioidessa.

Laitathan kännykän äänettömälle testin ajaksi.

Jos kokeen aikana tulee kysyttävää, voit soittaa Sampalle 0440321069.

APPENDIX D: Matlab code for 7.8 khz filter

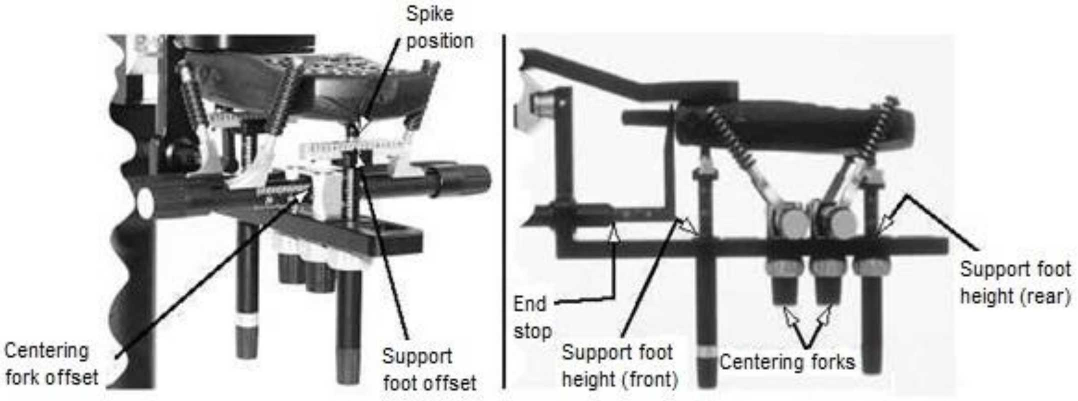
```
% 8kHz lowpass filter parameters (FIR equiripple)
% All frequency values are in Hz.

Fs = 48000;       % Sampling Frequency

Fpass = 7700;           % Passband Frequency
Fstop = 8300;           % Stopband Frequency
Dpass = 0.057501127785;  % Passband Ripple
Dstop = 0.001;          % Stopband Attenuation
dens  = 20;             % Density Factor

% Calculate the order from the parameters using FIRPMORD.
[N, Fo, Ao, W] = firpmord([Fpass, Fstop]/(Fs/2), [1 0], [Dpass, Dstop]);

% Calculate the coefficients using the FIRPM function.
b  = firpm(N, Fo, Ao, W, {dens});

% Sound file is read
[x, fs, nbits] = wavread('input_file.wav');

% Sound file is filtered using filter specs above
y = filter(b,1,x);

% The frequency response of original sound file is calculated
xF = 20*log10(abs(fft(x)));

% The frequency response of original sound file is plotted
figure(2);
plot(xF,'g');

% Hold is done to plot both curves to same figure
hold on;

% The frequency response of filtered sound file is calculated
yF = 20*log10(abs(fft(y)));

% The frequency response of filtered sound file is plotted
plot(yF,'r');

% Filtered sound file is written to hard drive
wavwrite(y,Fs,nbits,'output_file');
```

APPENDIX E: Matlab code for 5.5 khz lowpass filter

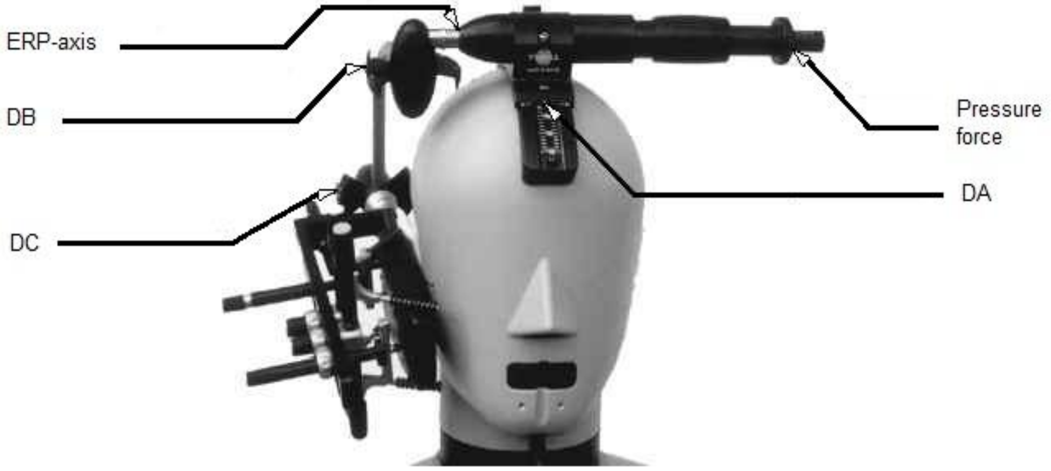
```
% 5,5kHz lowpass filter parameters.
% All frequency values are in Hz.

Fs = 48000;  % Sampling Frequency

Fpass = 5500;           % Passband Frequency
Fstop = 6400;           % Stopband Frequency
Dpass = 0.057501127785;  % Passband Ripple
Dstop = 0.001;          % Stopband Attenuation
dens  = 20;             % Density Factor

% Calculate the order from the parameters using FIRPMORD.
[N, Fo, Ao, W] = firpmord([Fpass, Fstop]/(Fs/2), [1 0], [Dpass, Dstop]);

% Calculate the coefficients using the FIRPM function.
b  = firpm(N, Fo, Ao, W, {dens});

% Sound file is read
[x, fs, nbits] = wavread('input_file.wav');

% Sound file is filtered using filter specs above
y = filter(b,1,x);

% The frequency response of original sound file is calculated
xF = 20*log10(abs(fft(x)));

% the frequency response of original sound file is plotted
figure(2);
plot(xF,'g');
hold on;

% the frequency response of filtered sound file is calculated
yF = 20*log10(abs(fft(y)));

% the frequency response of filtered sound file is plotted
plot(yF,'r');

% Filtered sound file is written to hard drive
wavwrite(y,Fs,nbits,'output_file');
```

APPENDIX F: Type 4606 hats position table for Nokia 6720c

End stopper			**Support foot**	Front	Rear
Endstop [mm]	13		Height[mm]	8	7
			Offset [-5…5]	0	0
Centering fork	Front	Rear	Socket position [1-5]	Front	4
Offset [mm]	0	0	Spike [Position]	+6,-6	+6,-6
Socket position [1-5]	1	5	Spike [Type]	long	long



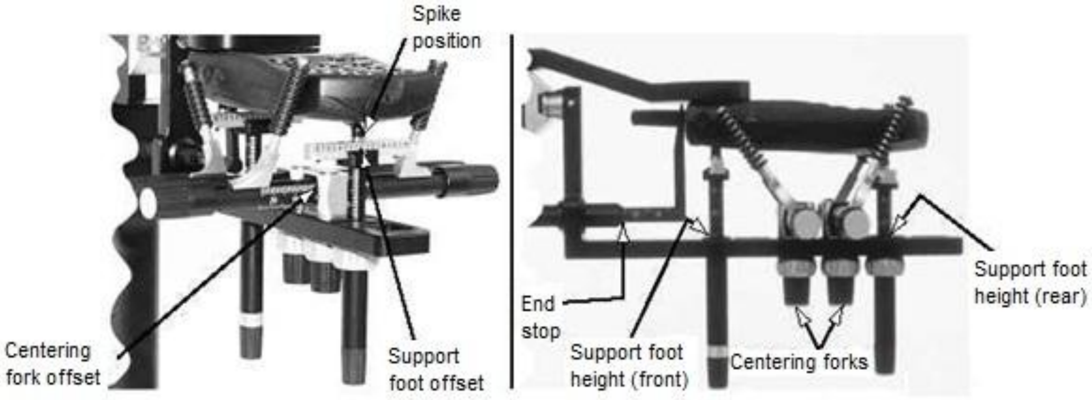
Position			Miscellaneous	
DA [°]	DB [°]	DC [°]	Pinna type	Pressure force [N]
21	10	2,5	Soft	10



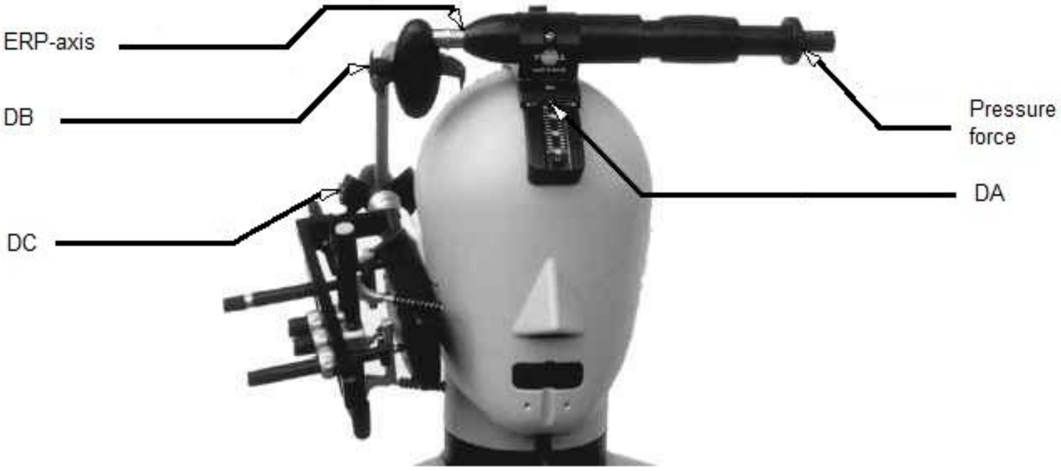
Nominal volume for Narrowband 6, Wideband 6

APPENDIX G: Type 4606 hats position table for Nokia 6220c

End stopper			**Support foot**	Front	Rear
Endstop [mm]	16		Height[mm]	8	8
			Offset [-5…5]	0	0
Centering fork	Front	Rear	Socket position [1-5]	Front	4
Offset [mm]	0	0	Spike [Position]	+6,-6	+6,-6
Socket position [1-5]	2	3	Spike [Type]	long	long



Position			Miscellaneous	
DA [°]	DB [°]	DC [°]	Pinna type	Pressure force [N]
21	10	-2	Soft	10



Nominal volume for Narrowband 5, Wideband 4