Chapter 5

Fulfillment of Desired Features of Sitar in Utility of Live Concerts of Contemporary Music

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In this chapter, the researcher will show how the features which have been described in the previous chapter can be fulfilled. For this we will focus on fulfillment of the electronic features.

Researcher will consider the following **Electronic Features**.

- **Amplification**
- **Changeable tonal Quality**
- **Changeability of sustention of the sound**
- **Tuning**
- **Recording of sound**
- **Retrieval of sound**
- **Monitoring of output sound**
- **Volume Control**
- **Mute**

5.1 Block Diagram.

Now focus on the below Block Diagram.

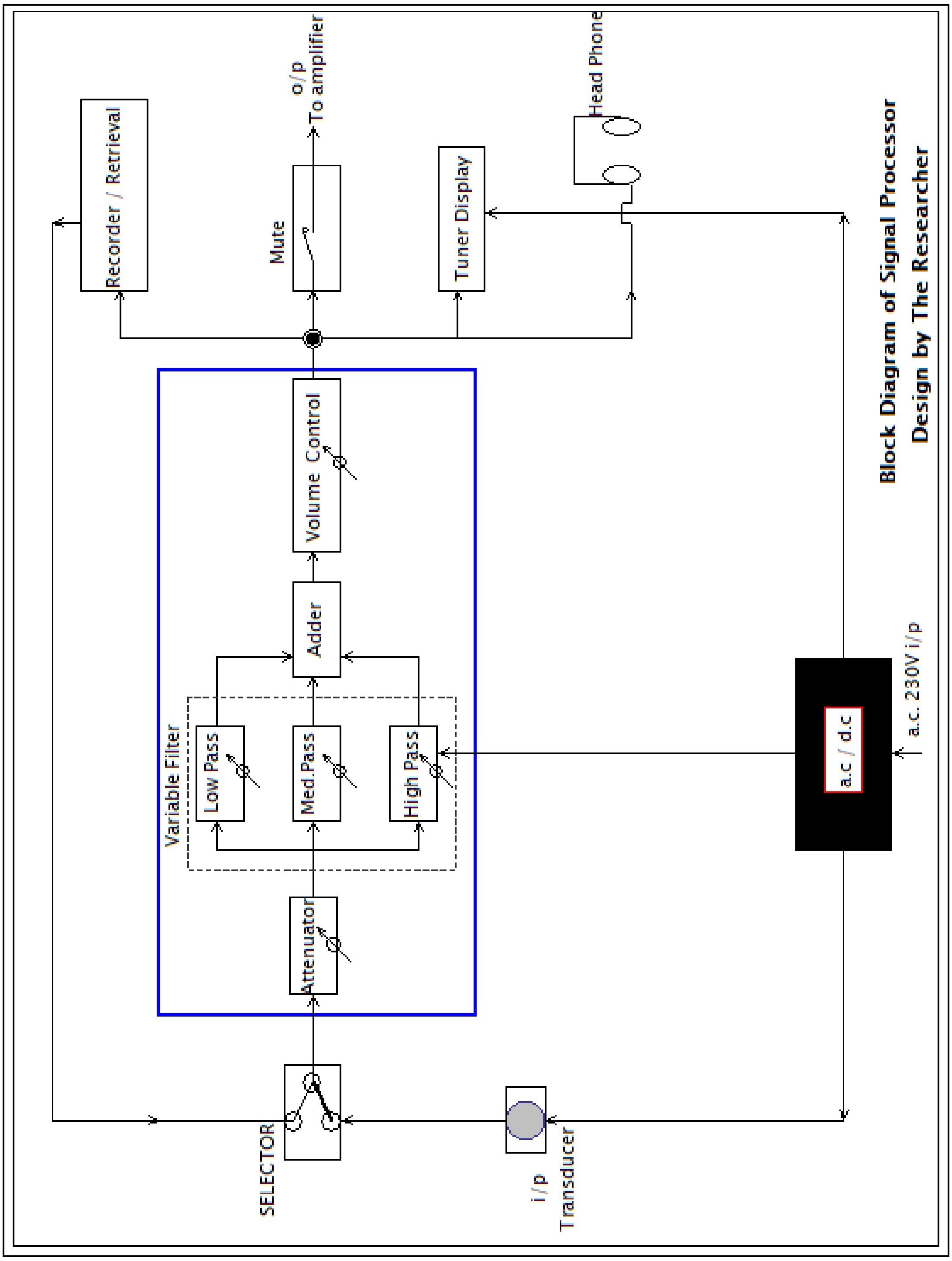

Fig 5.1 Block Diagram of Signal Processor

5.2. Fulfillment of Electronic Features

5.2.1 Amplification

To fulfil the first feature amplification:

Amplification of sound of Sitar is not exactly similar to that of the vocal sound. When amplification is done for the sound of Sitar, it is seen that the sound does not stay natural. The amplified sound loses its sweetness with its naturality. To avoid this factor a unit is designed as shown in the diagram consisting of the various blocks.

In the first step the sound wave is converted into the electrical signal and then it is sent for the amplification.

5.2.1.1 Transducer

We have to convert the sound into the electrical signal, in order to process and use the electrical methods and techniques to measure, manipulate, amplify, and control it.

One definition states[1] “A transducer is a device which, when actuated by energy in one transmission system, supplies energy in the same form or another form to second transmission system”. The energy transmission may be electrical, mechanical, chemical, optical (radiant), or thermal.

A unit or the device which converts one form of energy into another form of energy is known as the transducer. In our case it means the conversion of sound energy into electrical one.

There are two familiar types of the transducer as far as the sound wave is concerned.

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[1] Electronic Instrumentation and measurement techniques by W.D Cooper and A.D.Helfrick page 348

- Microphone and
- Pickup

The microphones are universally used for vocal music as well as for the instrumental music, but pickups are used only for the instrumental music.

5.2.1.1.1 Microphones

The most common type of the transducer is the microphone nicknamed Mic or Mike. A microphone[2] is an electroacoustic transducer actuated by energy in an acoustic system and delivering energy to an electrical system, the wave form in the electrical system being substantially equivalent to that in the acoustic system.

Microphones are used in many applications such as telephones, hearing aids, public address systems for concert halls and public events, live and recorded audio engineering, sound recording, two-way radios, radio and television broadcasting, and in computers for recording voice, speech recognition, and for non-acoustic purposes such as ultrasonic sensors or knock sensors.

Use of type of microphone depend on its application and preciseness required into that.

All microphone work[3] on the general basic principle that the energy of the sound wave is converted firstly into mechanical energy and then into electrical energy. So they all need a diaphragm which will vibrate when the sound wave produces a difference of pressure between its faces, and some means whereby mechanical movement can cause electrical signals to be generated. It is interesting to note that this is opposite to the action of a loudspeaker, where electrical energy is fed in to produce the mechanical

[2] Elements of acoustical engineering by Harry f. Olson page 172
[3] Acoustics by G.W. Mackenzie, first publication, page 80

vibrations which cause sound waves to be set up in the air. The microphone is an example of an electrical generator, while the loud speaker is an example of an electrical motor.

5.2.1.1.1.1 Types of Microphone

Microphones are categorized by their transducer principle, such as condenser, dynamic, etc.; and by their directional characteristics. Sometimes other characteristics such as diaphragm size, intended use or orientation of the principal sound input to the principal axis (end- or side-address) of the microphone are used to describe the microphone.

Several different types of microphone[4] are in use, which employ different methods to convert the air pressure variations of a sound wave to an electrical signal.

The most common are:

Carbon Microphone : Also known as a carbon button microphone, uses a capsule or button containing carbon granules pressed between two metal plates.

Dynamic Microphone : which uses a coil of wire suspended in a magnetic field.

Condenser Microphone : This uses the vibrating diaphragm as a capacitor plate.

Piezoelectric Microphone : which uses a crystal of piezoelectric material.

Laser Microphone: This microphone is a device that uses a laser beam and smoke or vapor to detect sound vibrations in free air.

Fiber optic Microphone: A fiber optic microphone converts acoustic waves into

[4] https://en.wikipedia.org/wiki/Microphone

electrical signals by sensing changes in light intensity, instead of sensing changes in capacitance or magnetic fields as with conventional microphones.

Speakers as Microphones: A loudspeaker, a transducer that turns an electrical signal into sound waves, is the functional opposite of a microphone.

Microphones, typically need to be connected to a preamplifier before the signal can be recorded or reproduced**.**

5.2.1.1.1.1.1 Carbon Microphone[5]

In carbon microphone, carbon granules are packed in to a box with the diaphragm as one side. When the diaphragm vibrates, the granules are subjected to a varying pressure, so that the areas of contact between the granules alters and the resistance of the device to electric current will change. Thus, if a current is passed through the carbon granules from a battery, the varying resistance will cause vibrations in the current and these vibrations are an electrical replica of the variations in pressure in the sound wave. The carbon microphone was used in the early days of recording and broadcasting but its quality is now no longer acceptable. Carbon microphones have **extremely low-quality sound reproduction and a very limited frequency response range, but are very robust devices, and economical in rates, so they are used where quality is not much of importance, and economy is required.**

5.2.1.1.1.1.2 Dynamic Microphone

Dynamic microphone (moving-coil microphone) works on principle of electromagnetic induction. They are robust, relatively inexpensive and resistant to

[5] Acoustics by G.W. Mackenzie, first publication, page 81

moisture. This, coupled with their potentially high gain before feedback, makes them **ideal for on-stage use**.

Dynamic microphones use the same dynamic principle as in a loudspeaker, only reversed. A small movable induction coil, positioned in the magnetic field of a permanent magnet, is attached to the diaphragm. When sound enters through the windscreen of the microphone, the sound wave moves the diaphragm. When the diaphragm vibrates, the coil moves in the magnetic field, producing a varying current in the coil through electromagnetic induction. A single dynamic membrane does not respond linearly to all audio frequencies. For this reason some microphones utilize multiple membranes for the different parts of the audio spectrum and then combine the resulting signals. Combining the multiple signals correctly is difficult and designs that do this are rare and tend to be expensive. On the other hand there are several designs that are more specifically aimed towards isolated parts of the audio spectrum. The AKG D 112, for example, is designed for bass response rather than treble. In audio engineering several kinds of microphones are often used at the same time to get the best results.

5.2.1.1.1.1.3 Condensor Microphone

Condenser microphone, is also called a **Capacitor Microphone or Electrostatic Microphone**, as capacitors were called condensers. Here, the diaphragm acts as one plate of a capacitor, and the vibrations produce changes in the distance between the plates. There are two types, depending on the method of extracting the audio signal from the transducer: DC-biased microphones, and radio frequency (RF) or high frequency (HF) condenser microphones. With a DC-biased microphone, the plates are biased with a fixed charge (Q). The voltage maintained across the capacitor plates changes with the vibrations in the air, according to the capacitance equation

$$\mathbf{(C = Q/V),}$$

Where Q = Charge in Coulombs,

C = Capacitance in Farads and

V = Potential difference in Volts.

The capacitance of the plates is inversely proportional to the distance between them for a parallel-plate capacitor. The assembly of fixed and movable plates is called an 'Element' or 'Capsule'.

A nearly constant charge is maintained on the capacitor. As the capacitance changes, the charge across the capacitor does change very slightly, but at audible frequencies it is sensibly constant. The capacitance of the capsule (around 5 to 100 pF) and the value of the bias resistor (100 MΩ to tens of GΩ) form a filter that is high-pass for the audio signal, and low-pass for the bias voltage. Note that the time constant of an RC circuit equals the product of the resistance and capacitance.

Within the time-frame of the capacitance change (as much as 50 ms at 20 Hz audio signal), the charge is practically constant and the voltage across the capacitor changes instantaneously to reflect the change in capacitance. The voltage across the capacitor varies above and below the bias voltage. The voltage difference between the bias and the capacitor is seen across the series resistor. The voltage across the resistor is amplified for performance or recording. In most cases, the electronics in the microphone itself contribute no voltage gain as the voltage differential is quite significant, up to several volts for high sound levels. Since this is a very high impedance circuit, current gain only is usually needed, with the voltage remaining constant.

Fig. 5.2 AKG C451B small-diaphragm condenser microphone

RF condenser microphones use a comparatively low RF voltage, generated by a low-noise oscillator. The signal from the oscillator may either be amplitude modulated by the capacitance changes produced by the sound waves moving the capsule diaphragm, or the capsule may be part of a resonant circuit that modulates the frequency of the oscillator signal. Demodulation yields a low-noise audio frequency signal with a very low source impedance. The absence of a high bias voltage permits the use of a diaphragm with looser tension, which may be used to achieve wider frequency response due to higher compliance. The RF biasing process results in a lower electrical impedance capsule, a useful by-product of which is that RF condenser microphones can be operated in damp weather conditions that could create problems in DC-biased microphones with contaminated insulating surfaces. The Sennheiser 'MKH' series of microphones use the RF biasing technique.

Condenser microphones span the range from telephone transmitters through inexpensive karaoke microphones to high-fidelity recording microphones. They generally produce a high-quality audio signal and are now the popular choice in laboratory and recording studio applications. The inherent suitability of this technology is due to the very small mass that must be moved by the incident sound wave, unlike other microphone types that require the sound wave to do more work. They require a power source, provided either via microphone inputs on equipment as phantom power or from a small battery. Power is necessary for establishing the capacitor plate voltage, and is also needed to power the microphone electronics (impedance conversion in the case of electret and DC-polarized microphones, demodulation or detection in the case of RF/HF microphones). Condenser microphones are also available with two diaphragms that can be electrically connected to provide a range of polar patterns.

5.2.1.1.1.1.4 Piezoelectric Microphone or Crystal Microphone[6]

A Crystal microphone sometimes referred as Piezo microphone uses the phenomenon of piezoelectricity. If certain materials are mechanically deformed, i.e. bent or twisted, a difference of voltage is produced between the faces of the material. This is known as the piezo electric effect and is obviously applicable to the microphones since we can use the forces due to a sound wave to drive a thin layer of suitable material. Although there are several possible materials, the one normally used is **Rochelle salt** since this is very sensitive and produces a good output signal. The high impedance of the crystal

[6] Acoustics by G.W. Mackenzie, first publication, page 83

microphone made it very susceptible to handling noise, both from the microphone itself and from the connecting cable.

Piezoelectric transducers are often used as contact microphones to amplify sound from acoustic musical instruments, to sense drum hits, for triggering electronic samples, and to record sound in challenging environments, such as underwater under high pressure. Saddle-mounted pickups on acoustic guitars are generally piezoelectric devices that contact the strings passing over the saddle. This type of microphone is different from magnetic coil pickups commonly visible on typical Sitar, electric guitars, which use magnetic induction, rather than mechanical coupling, to pick up the vibrations.

Phantom power in fact, this voltage can damage some older ribbon microphones. Some new modern ribbon microphone designs incorporate a preamplifier and, therefore, do require phantom power, and circuits of modern passive ribbon microphones, i.e., those without the aforementioned preamplifier, are specifically designed to resist damage to the ribbon and transformer by phantom power. Also there are new ribbon materials available that are immune to wind blasts and phantom power.

5.2.1.1.1.1.5 Laser Microphone

Laser microphones are often portrayed in movies as spy gadgets, because they can be used to pick up sound at a distance from the microphone equipment. A laser beam is aimed at the surface of a window or other plane surface that is affected by sound. The vibrations of this surface change the angle at which the beam is reflected, and the motion of the laser spot from the returning beam is detected and converted to an audio signal.

In a more robust and expensive implementation, the returned light is split and fed to an interferometer, which detects movement of the surface by changes in the optical path length of the reflected beam. The former implementation is a tabletop experiment; the latter requires an extremely stable laser and precise optics.

A new type of laser microphone is a device that uses a laser beam and smoke or vapor to detect sound vibrations in free air.

5.2.1.1.1.1.6 Fiber Optic Microphone:

Fig. 5.3 Optoacoustics Fiber Optic Microphone

During operation, light from a laser source travels through an optical fiber to illuminate the surface of a reflective diaphragm. Sound vibrations of the diaphragm modulate the intensity of light reflecting off the diaphragm in a specific direction. The modulated light is then transmitted over a second optical fiber to a photo detector, which transforms the intensity-modulated light into analog or digital audio for transmission or recording. Fiber optic microphones possess high dynamic and frequency range, similar to the best high fidelity conventional microphones. Fiber optic microphones do not react to or influence any electrical, magnetic, electrostatic or radioactive fields (which is known as EMI/RFI immunity). The fiber optic microphone design is therefore ideal for use in areas where conventional microphones are ineffective or dangerous, such as inside industrial application or in magnetic resonance imaging (MRI) equipment environments.

Fiber optic microphones are robust, resistant to environmental changes in heat and moisture, and can be produced for any directionality or impedance matching. The distance between the microphone's light source and its photo detector may be up to several kilometers without need for any preamplifier or other electrical device, making **Fiber Optic Microphones suitable for Industrial and Surveillance Acoustic Monitoring.**

Fiber optic microphones are used in very specific application areas such as for infrasound monitoring and noise-canceling. They have proven especially useful in medical applications, such as allowing radiologists, staff and patients within the powerful and noisy magnetic field to converse normally, inside the MRI suites as well as in remote control rooms. Other uses include industrial equipment monitoring and audio calibration and measurement, high-fidelity recording and law enforcement.

5.2.1.1.1.1.7 Speakers as Microphones

A loudspeaker, a transducer that turns an electrical signal into sound waves, is the functional opposite of a microphone. Since a conventional speaker is constructed much like a dynamic microphone (with a diaphragm, coil and magnet), speakers can actually work ‘in reverse’ as microphones. The resulting signal typically offers reduced quality including limited high-end frequency response and poor sensitivity. In practical use, speakers are sometimes used as microphones in applications where high quality and sensitivity are not needed such as intercoms, walkie-talkies or video game voice chat peripherals, or when conventional microphones are in short supply. However, there is at least one practical application that exploits those weaknesses: the use of a medium-size woofer placed closely in front of a ‘Kick Drum’ (Bass Drum) in a drum set to act as a microphone. Since a relatively massive membrane is unable to transduce high frequencies while being capable of tolerating strong low-frequency transients, the speaker is often ideal for picking up the kick drum while reducing bleed from the nearby cymbals and snare drums. Less commonly, microphones themselves can be used as speakers, but due to their low power handling and small transducer sizes, a tweeter is the most practical application.

5.2.1.1.1.1.8 MEMS Microphones

The Meaning of MEMS is Micro Electrical-Mechanical System. This microphone is also called a microphone chip or silicon microphone. It is one of the latest kind of microphone. A pressure-sensitive diaphragm is etched directly into a silicon wafer by MEMS processing techniques, and is usually accompanied with integrated preamplifier. Most MEMS microphones are variants of the condenser microphone design. Digital MEMS microphones have built in analog-to-digital converter (ADC) circuits on the same CMOS chip making the chip a digital microphone and so more readily integrated with modern digital products.

5.2.1.1.1.2 Factors to be considered while Selecting Microphone

5.2.1.1.1.2.1 Microphones Out-of-Doors[7]

Wind noises are a great source of trouble with outdoor broadcasting or recording. Normally, ribbon microphones are useless out-of-doors, as the ribbon has to be fairly loosely tensioned and can be easily moved by the wind. Moving coil and crystal microphones are suitable for outdoor use. If the wind noise is troublesome form of wind shielding should be tried. Proper windshields are in fact supplied for some microphones, and this generally consists of layers of porous material with shield the microphone from the air currents but allow the sound wave to reach the diaphragm more or less unaltered. It is a good idea to insert some bass cut in the microphone circuit, as this will attenuate the turbulence tones

5.2.1.1.1.2.2 Use for Speech or Music / Intelligibility Required[8]

One has to understand the characteristics of the speech and the music to see how it imposes certain requirement while selecting the microphone.

When listening to another person speaking, one is conscious that the voice has a characteristic pitch. A male voice has a lower pitch than a female and in turn the female voice is of a lower pitch than a child's. The average pitch for a man is about 130 c/s; that for a woman is about twice this. It is obvious that the actual pitch varies about this average values depending upon the individual. The fluctuating air stream leaving the mouth has a complex wave form. As it is known, this will consist of a fundamental note along with its harmonics. It is the frequency of this fundamental which determines

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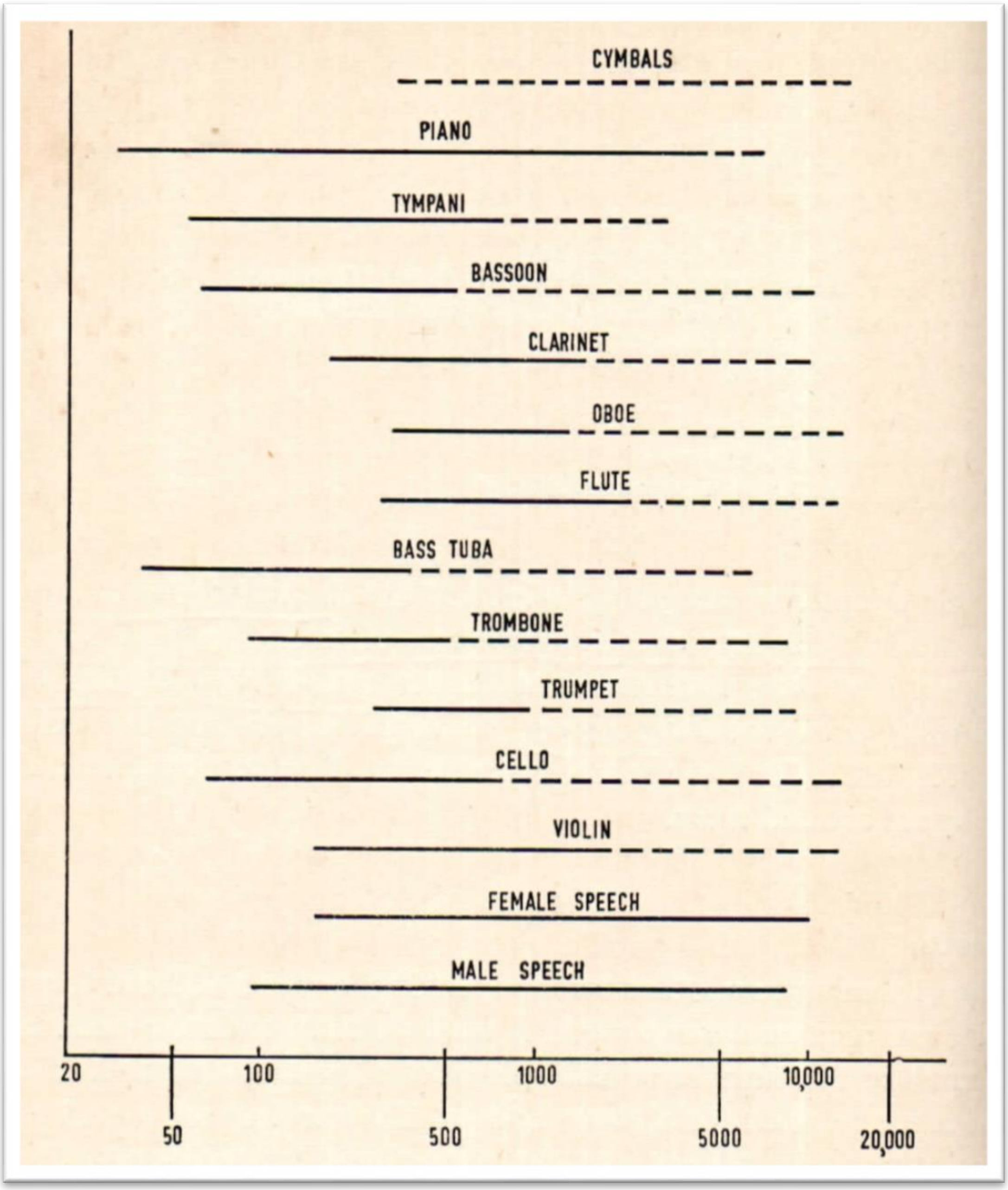
[7] Acoustics by G.W. Mackenzie, first publication, page 121

[8] Acoustics by G.W. Mackenzie, first publication, page 61

the pitch of the voice. The cavities in head resonate to accentuate particular groups of harmonics in the fluctuating air stream. These frequencies are called the formants. Varying the size of these cavities by moving the tongue, jaw, lips, etc. alters the resonant frequency for the various speech sounds. Natural male speech has an overall frequency range of from 100-8000 c/s; for woman the range is something like 200-10000 c/s. within these ranges there are some frequencies which are more important than others to the understanding of what is being said.



Fig[9]. 5.4 The frequency ranges of speech and some of the more common musical instruments. The solid line indicates the fundamental notes and the dotted extensions the harmonics

Most of the energy in the speech is contained in the low frequencies but they contribute very little to the understanding of the speech. To put in another way,

[9] Acoustics by G.W. Mackenzie, first publication, page 62

these frequency do not control the 'Intelligibility' of the speech. When one hears only the low frequencies, say only up to 500 c/s, the speech sound muffled and woolly and it is impossible to make any sense out of the speech.

It is the high frequencies which contain the intelligence. Listening to speech which has a severely restricted based, one can still easily understand what is being said although of course it lacks power and sounds 'thin'.

Music has a much wider frequency and intensity range than speech. From a frequency point of view, a high fidelity reproduction system for music should be capable of handling a range from 30 c/s to 15000 c/s.

The maximum intensity range of music directly heard, is of the order of 70 dB. This is too wide for most recording or broadcasting systems and some compression is necessary. The frequency range just quoted is of course for the entire range of sounds produced by the musical instruments which are normally used. The range can be broken down, first of all in two the individual ranges of the various instruments and then, for a particular instrument, in to the range of the fundamental notes and overtones it produces.

Most musical notes are complex, consisting of fundamental and a series of harmonics. With the oboe, the range of the fundamental notes it produces is from 233c/s to 1574 c/s approximately. The harmonics extend this range to about 11000 c/s. The violin has a fundamental range of about 1000 c/s starting with its lowest string at 200 c/s. with the harmonics, the highest frequency produced is round about 14000 c/s. It is the same with all the sources of musical sounds, the human voice and the instruments. And to hear these sources properly, not only must all the frequencies- fundamentals and harmonics- be present, but they must retain their correct amplitudes relative to each other. So much for frequency; from an intensity range point of view, two factors are important. The maximum range which is already seen is 70 db. In practice, much music does not exploit this full intensity range; the range depends on how the individual composer has used the instruments at his disposal.

The cost of the microphone depends on the frequency bandwidth which it offers. So microphone should be selected depending upon its usage in speech or in the music.

5.2.1.1.1.2.3 Directivity of Microphone[10]

The graph of a microphone's response to sounds coming from different angle is called a polar diagram or directivity pattern. This is obviously important as, in most cases, sound reach the microphone not only by the direct path but by reflected paths from the walls of the room, etc. The shape of its directivity pattern has a great bearing on how a microphone should be placed so that an acceptable balance between direct and reflected sound is obtained.

With known directional properties Microphone can be angled in such a way that the contribution from undesired source can be considerably attenuated.

Ideally, of course, the directivity pattern, whatever its shape, should be the same at all frequencies in the working range of the microphone. However, in practice this is somewhat difficult to achieve because of the large range of wavelength a microphone has to handle specially in case of the music (Compare to speech). As a result very few microphones have directivity patterns which are completely independent of frequency.

It is found that at low frequencies-long wave lengths-the distribution is almost uniform. With the high frequencies-smaller wave length-the sound becomes

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[10] Acoustics by G.W. Mackenzie, first publication, page 67

concentrated over a smaller frontal region. The same general distribution pattern is produced by musical instruments. When the wave length is large compared with the source dimensions, there is a very little concentration or beaming of energy. As frequency rises the wave length gets smaller. When it approaches or gets smaller than the source dimensions, the distribution becomes directional markedly so in some cases.

Omni-directional: The simplest pattern of course a circle, indicating that the microphone is equally sensitive in all directions- The so called Omni- Directional pattern. The directivity pattern of a pressure operated microphone tends to be omni directional at low frequencies and gradually becomes increasingly directional as the frequency rises.

High- Directional microphone: There are many situations where microphone can not be placed near the source of sound, for various reasons. These usually occur out of doors; bird songs, the crack of the ball on the cricket bat, the commands from the officer at a military parade are some examples. For these situations it is essential to have a microphone which picks up sound energy only over a small angle or to have in other words, high directivity.

5.2.1.1.1.2.4 Sensitivity of Microphone[11]

The electrical output of a microphone for a given sound strength should be as high as possible, so that an adequate signal-to-noise ratio is obtained at the start of the electrical chain from microphone to amplifier or to the loud speaker. The output voltage of the microphone will depend, in the first instance on how much of the sound wave affects the diaphragm. This must depend on the size- more strictly the area -of the diaphragm and so it can not be reduced too far, otherwise the sensitivity of the

[11] Acoustics by G.W. Mackenzie, first publication, page 95

microphone will be too low. So as per this theory size of the microphone can not be reduced beyond certain level. But today because of some latest versions of the technology button sort of microphones are also available, which are used mostly in the on stage applications.

5.2.1.1.1.2.5 Impedance of Microphone[12]

The output impedance of a microphone must be known before it is connected to an amplifier or a speaker. In practice, the impedance of a microphone can range from a few ohms to several thousand ohms.

Moving coil and ribbon microphones are low impedance generator, typical figure being about 30 ohms and 1 ohm, respectively.

While Crystal microphones have a very high output impedance

With the ribbon, a transformer is normally built inside the case to set up the impedance, the turn ratio of the transformer controlling the output impedance. Some models have fixed impedances, typical values being 30 ohms or 300 ohms. With other types, a choice of impedance is available and this can be Low-25 ohm, Line-600 ohms or High-50000 Ohms.

The advantage of having a low impedance microphone is that the length of cable connecting it to the rest of the equipment can be as long as one wishes, 100 metre or so. The disadvantage is that a transformer must be used if one has equipment with high input impedance, to match the microphone and its cable into the equipment. In addition the sensitivity of low impedance microphone is generally low.

[12] Acoustics by G.W. Mackenzie, first publication, page 96

All high impedance, low capacity cables are susceptible to hum and interference pickup and can generate noise signals if they are moved. It is therefore essential to take care with the cables of high impedance microphones.

5.2.1.1.1.2.6 Placement of Microphone for Musical Instruments[13]

Also note that musical instruments do not distribute the sound energy equally at all frequencies. High frequencies tend to become beamed, and contained in this frequencies are the upper harmonics which contribute a great deal to the individual instrument's tonal quality.

This is obviously a most important factor to be taken into account when recording a solo instrument. It is not the only one however; not only do instrument produce the required musical notes, but they also produce noises due to their mechanical action. Therefore care must be taken **in placing** the microphone to get a blend of satisfactory tone quality with the minimum of noise.

- For the **String Instrument** family the violin has the most marked beaming of the higher frequencies, and the microphone should be placed at right angles to the front face of the instruments initially and then moved to one side perhaps until the noise is reduced. Obviously trial and error is called for, until the balance is right. Of the other stringed instruments, it is important to let the cello 'see' the microphone. With its low playing position it is easy to lose the instrument among other instruments. With the double bass, one can often hear sounds which are just thumps instead of being clearly defined. This is more than likely due to the instrument causing resonances in the surrounding, and

[13] Acoustics by G.W. Mackenzie, first publication, page 132

considerable improvement is possible by standing the instrument on a solid floor.

- For **Brass Instrument:** The trumpet and the trombone are good examples of highly directional sound sources. Unlike the string instrument, however, they have a large output, so the microphone should be placed well away, say about 5 ft. If the player tends to move while playing, it is safer to place the microphone to one side of the axis of the instrument. The loss of some of the upper harmonics will not then be noticeable.
- For the **Piano type Instruments:** A great deal must obviously depend on whether the instrument is a grand or an upright, what type of music is being played and what the acoustic surroundings are like. With an upright piano, a reasonable balance is obtained with a bidirectional microphone.
- For **Percussion Instruments:** Even in professional work the microphone placement for these instruments is difficult since they tend to be powerful and called for a distant microphone to cope with the volume. However this means that the result is ‘blurred’ or lacking in definition. So with drums and cymbals various microphone positions should be used until a satisfactory compromised is obtained.

The microphones which are studied here are concerned with the musical instruments. There are various kind of the instruments as follows.

- String Instrument
- Wind Instruments
- Percussion Instruments
- Solid Instruments
- Electronic Instruments.

The microphones used for different type of instruments are not same. This is a very important thing to notice. Different type of the mics are suitable for different instruments.

In Short Microphone should be Chosen Depending Upon

- Its indoor or outdoor usage
- Intelligibility required. (i.e. It is used either for speech or music)
- Type of musical instrument in case of music.
- Precision required versus cost.

Below attached sheet contains a variety of indoor and outdoor microphones and pickups useful for vocal and instrumental music.

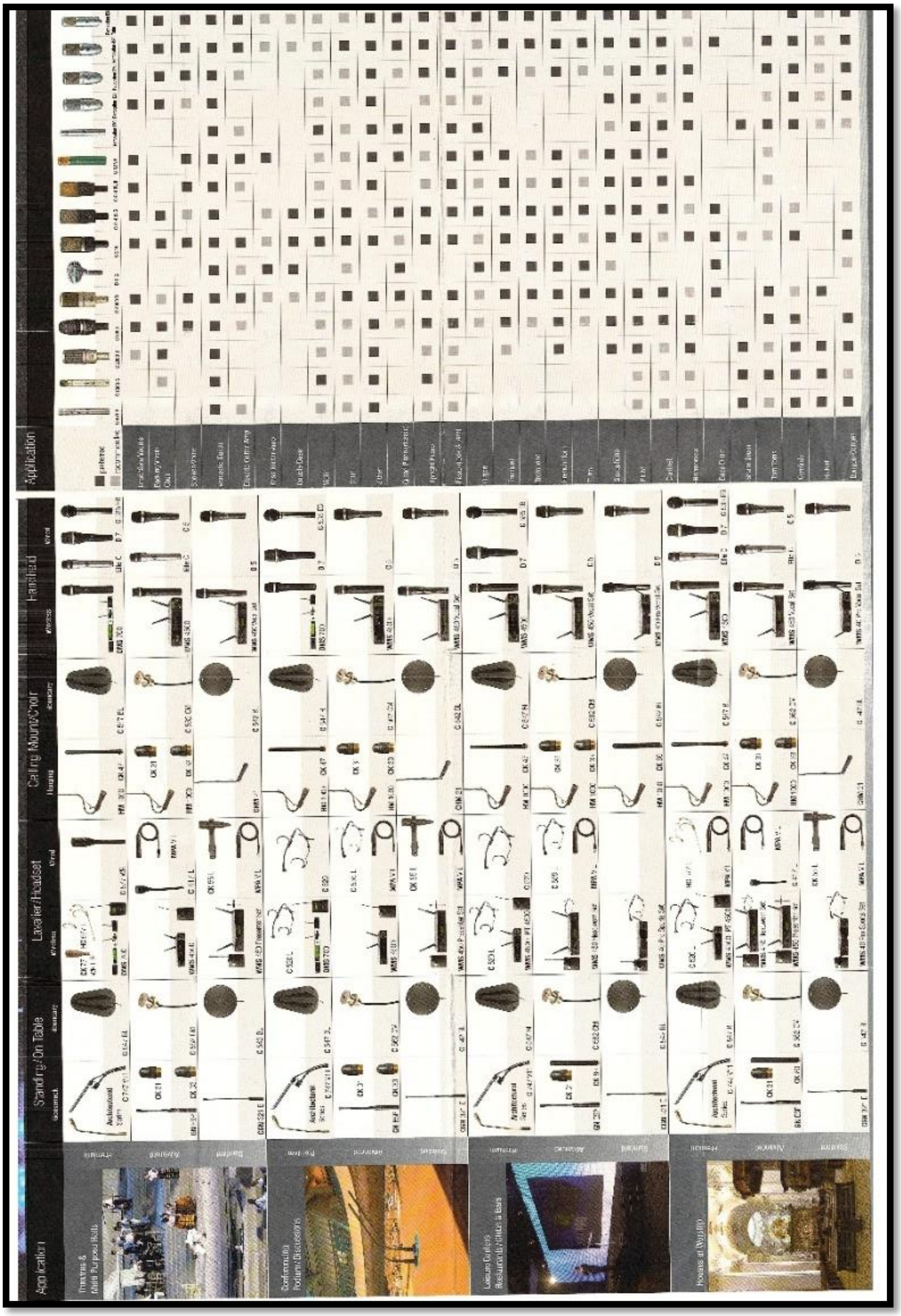


Fig 5.5 Various Types of the Mics

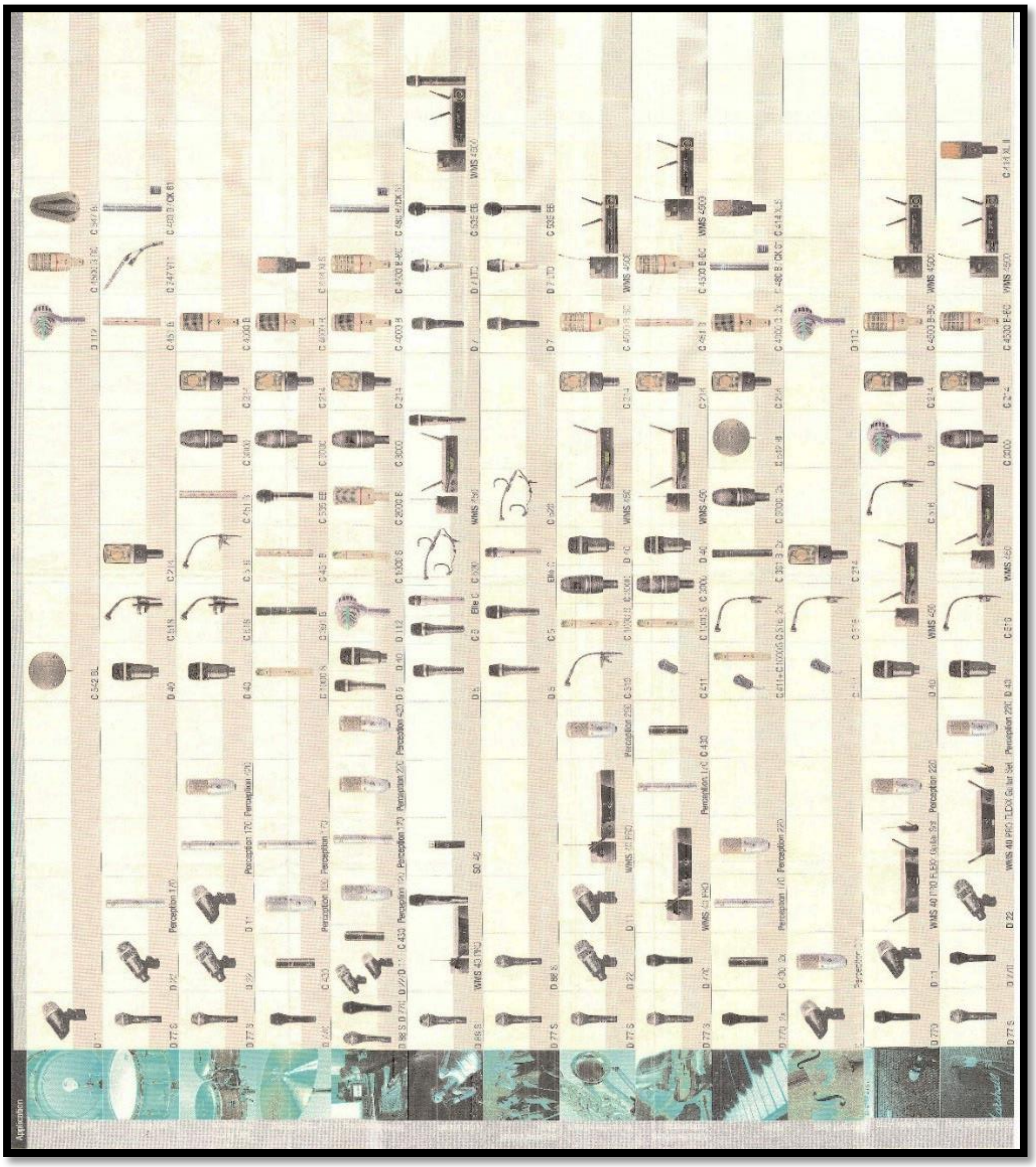


Fig .5.6 Various Types of the Mics

5.2.1.1.2 PICK UP[14]

A **Pickup** device is a type of transducer (specifically a variable reluctance sensor) that captures or senses mechanical vibrations produced by musical instruments, particularly stringed instruments such as the Sitar, electric guitar, electric bass guitar violin etc. and converts them to an electrical signal that is amplified using an instrument amplifier (such as preamp) to produce musical sounds through a loudspeaker in a speaker enclosure. Most electric guitars and electric basses use magnetic pickups. Sitar, acoustic guitars, upright basses and fiddles often use a piezoelectric pickup.

There are basically four principles used to convert sound into an alternating current, each with their pros and cons:

- A **Microphone** registers the vibrations of the air caused by the instrument. In general this technique guarantees a good sound quality, but with two limitations: feedback and crosstalk.
- **Contact Pickups** register the vibrations of the instrument itself. They have the advantage of producing little feedback and **no Cross-Talk at all**. In spite of their lesser sound quality and low price, contact pickups (and especially the piezoelectric pickup) have become the most popular transducer.
- **Magnetic Pickups**. Magnetic pickups, as applied in electric guitars, register the vibrations of nickel or steel strings in a magnetic field. They have the, but in combination with a steel-string acoustic guitar the sound

[14] https://en.wikipedia.org/wiki/Pickup_(music_technology)

tends to be electric. This is why acoustic guitarists typically choose a piezoelectric pickup, built in microphone, or both.

- **Electrostatic pickups**. Another way is to use the changing capacitance between the string and a pickup plate. These electronic pickups produce much higher dynamics than conventional pickups, so the difference between a soft and a loud pick strike is more pronounced than with other types of pickups

An amplification system with two transducers combines the qualities of both.

A combination of a microphone and a piezoelectric pickup typically produces better sound quality and less sensitivity to feedback, as compared to single transducers.

However, this is not always the case. A less frequently used combination is a piezoelectric and a magnetic pickup. This combination can work well for a solid sound with dynamics and expression. Examples of a double system amplifier are the Highlander iP-2, the Verweij VAMP or the LR Baggs dual source and the D-TAR Multisource.

5.2.1.1.2.1 Magnetic Pickups

A magnetic pickup is made up of the magnetic material like permanent magnet with a core of material such as alnico or ferrite, wrapped with a coil of several thousand turns of fine enamelled copper wire. The pickup is most often mounted on the body of the instrument, but can be attached to the bridge, neck or pick guard, as on many electro-acoustic arch top jazz guitars. Magnetic pickups used with string basses can be attached to the bridge. The permanent magnet creates a magnetic field; the motion of the vibrating steel strings disturbs the field, changing magnetic flux and inducing an electric current through the coil. The pickup is then connected with a patch cable to an amplifier which amplifies the signal to a sufficient magnitude of power to drive a loudspeaker. A pickup can also be connected to recording equipment via a patch cable. There may also be an internal preamplifier device mounted in an acoustic guitar or in an external box. When a preamp is used in this way, it is between the pickup and cable and can significantly reduce the equivalent impedance of the pickup coil.

Output

The output voltage of magnetic pickups varies between 100 mV r.m.s to over 1 V r.m.s for some of the higher output types. Some high-output pickups achieve this by employing very strong magnets, thus creating more flux and thereby more output. This can be detrimental to the final sound because the magnet's pull on the strings can cause problems with intonation as well as **Damp the Strings and reduce Sustain.** Other high-output pickups have more turns of wire to increase the voltage generated by the string's movement. However, this also increases the pickup's output resistance/impedance, which **can affect high frequencies** if the pickup is not isolated by a buffer amplifier or a DI unit.

The turns of wire in proximity to each other have an equivalent self-capacitance that, when added to any cable capacitance present, resonates with the inductance of the winding. This resonance can accentuate certain frequencies, giving the pickup a characteristic tonal quality. The more turns of wire in the winding, the higher the output voltage but the lower this resonance frequency. The inductive source impedance inherent in this type of transducer **makes it less linear than other forms of pickups,** such as piezo-electric or optical. The tonal quality produced by this

nonlinearity is, however, subject to taste, and some string instrumentalist and luthiers consider it aesthetically superior to a more linear transducer.

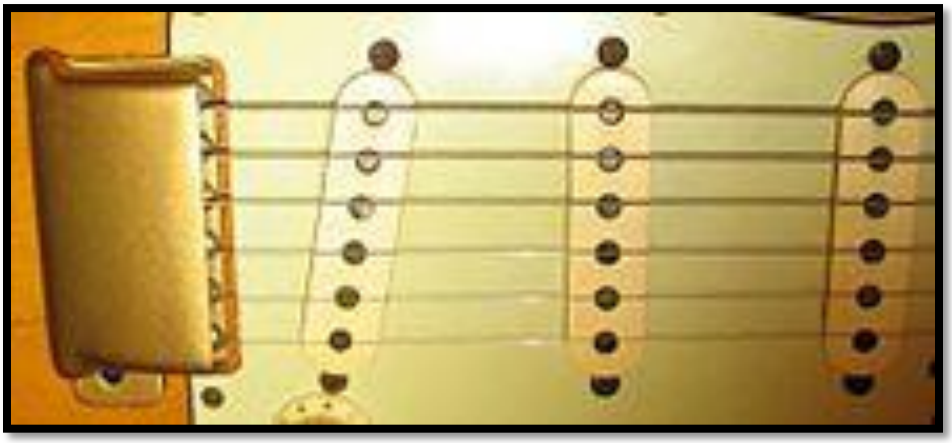


Fig. 5.7 Single Coil Pickups, Fender Stratocaster (1963)

The external load usually consists of resistance (the volume and one potentiometer in the string instrument, and any resistance to ground at the amplifier input) and capacitance between the hot lead and shield in the connecting cable. The electric cable also has a capacitance, which can be a significant portion of the overall system capacitance. This arrangement of passive components forms a resistively-damped second-order low-pass filter. Pickups are usually designed to feed a high input impedance, typically a Meg ohm or more, and a low impedance load reduces the high-frequency response of the pickup because of the filtering effect of the inductance.

5.2.1.1.2.2 Piezo Electric Pick up

Sometimes string instrument player fit piezoelectric pickups instead of, or in addition to, magnetic pickups. These have a very different sound, and also have the advantage of not picking up any other magnetic fields, such as mains hum and feedback from monitoring loops. In hybrid guitars, this system allows switching between magnetic pickup and piezo sounds, or simultaneously blending the output. Solid bodied guitars with only a piezo pickup are known as silent guitars, which are usually used for practicing by acoustic guitarists. **Piezo pickups** can also be built into electric guitar bridges for conversion of existing instruments.

Most pickups for bowed string instruments, such as violin, and double is piezoelectric. These may be inlaid into the bridge, laid between the bridge feet and the top of the instrument, or, less frequently, wedged under a wing of the bridge. Some pickups are fastened to the top of the instrument with removable special putty usually of the black colour.

This putty is also carrying special characteristics of passing the vibrations on the surface to the transducer. Putty can not be replaced by ordinary glue or other sticky material.

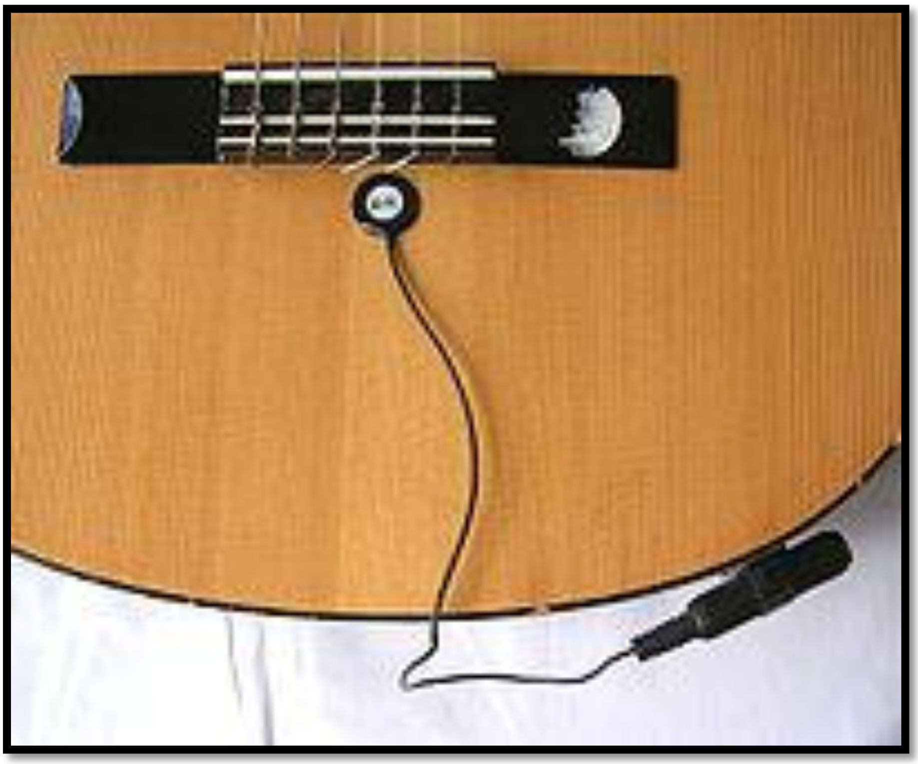


Fig. 5.8 Piezoelectric Pickup on a Classical Acoustic Guitar

Fig. 5.9 Dual Pick up by Peterman in Australia

Fig 5.10 Piezoelectric Violin Bridge Pickup

The Piezo Pickup gives a very wide frequency range output compared to the Magnetic types and can give large amplitude signals from the strings.

- **Preamps**

Piezoelectric pickups have a very high output impedance and appear as a capacitance in series with a voltage source. They therefore often have an instrument-mounted buffer amplifier fitted to maximize frequency response.

The buffer amplifier is often powered from relatively high voltage rails (about ±9 V) to avoid distortion due to clipping. A less linear preamp (like a single-FET amplifier) might be preferable due to softer clipping characteristics. Such an amplifier starts to distort sooner, which makes the distortion less 'buzzy' and less audible than a more linear, but less forgiving op-amp However, at least one study indicates that most people cannot tell the difference between FET and op-amp circuits in blind listening comparisons of electric instrument preamps, which correlates with results of formal studies of other types of audio devices. Sometimes, piezoelectric pickups are used in conjunction with magnetic types to give a wider range of available sounds.

5.2.1.1.2.3 Optical Pickup[15]

Optical pickups are a fairly recent development that work by sensing the interruption of a light beam by a vibrating string. The light source is usually an LED, and the detector is a photodiode or phototransistor. These pickups are completely resistant to magnetic or electric interference and also have a very broad and flat frequency response, unlike magnetic pickups.

Optical pickup guitars were first shown at the 1969 NAMM in Chicago, by Ron Hoag

In 2000, Christopher Willcox, founder of Light Wave Systems, unveiled a new beta technology for an optical pickup system using infrared light. In May 2001, Light Wave Systems released their second generation pickup, dubbed the "S2." The S2 featured Light Wave Systems' monolithic bridge, six-channel motherboard, and a host of other improvements, making the technology more practical for use in both live and recording studio settings.

[15] https://en.wikipedia.org/wiki/Pickup_(music_technology)

5.2.1.1.2.4 Multi-Transducer Pickups[16]

This type of the pickups are not used by artists, except some professionals. Sometimes this **Single Pickup** contains **Sub Six Pickups**. So it is called Hexaphonic pickup. Hexaphonic pickups also called **Divided Pickups** and **Polyphonic Pickups** have a separate output for each string (Hexaphonic assumes six strings). This allows for separate processing and amplification for each string. It also allows a converter to sense the pitch coming from individual string signals for producing note commands, typically according to the MIDI (musical instrument digital interface) protocol. A hexaphonic pickup and a converter are usually components of a guitar/synthesizer.

Such pickups are uncommon (compared to normal ones), and only a few notable models exist, like the piezoelectric pickups on the Moog Guitar. Hexaphonic pickups can be either magnetic or piezoelectric or based on the condensor principle like electronic pickups.

[16] https://en.wikipedia.org/wiki/Pickup_(music_technology)

5.2.1.1.2.5 Active and Passive Pickups

Pickups can be either active or passive. Pickups, apart from optical types, are inherently passive transducers. 'Active' pickups incorporate electronic circuitry to modify the signal.

Fig.5.11 EMG 81 and EMG 85 Pair of Popular Pickups

'**Passive**' pickups are usually wire wound around a magnet, and are the most common type used. They can generate electric potential without need for external power, though their output is relatively low, and the harmonic content of output depends greatly on the winding.

Fig. 5.12 Seymour Duncan AHB-1 Blackouts

Active pickups utilize the same type of reluctance sensor as a passive pickup (although features such as coil wire size and number of windings may vary from those used in a passive pickup). Active pickups require an electrical source of energy (usually one or two 9V batteries) to operate and include an electronic preamp very similar to the preamp or buffer found in most amplifiers and effects circuits. These circuits can be designed to give a large range of gain for a large range of possible output power. The circuitry is virtually identical to any preamplifier or buffer found in amplifiers and effects circuits. Unlike the preamp of an amplifier circuit, the Op Amps used need to be of a low power design to optimize battery life and they are unable to utilize a Rail to Rail input power configuration due to the low battery power. This limits the dynamic range of the circuit when compared to a Rail to Rail Op Amp preamp circuit as found in most amplifiers. By adding an additional preamplifier/buffer active pickups are able to filter attenuate or boost the signal from the pickup. Any extra voltage gain added to the output signal will increase clipping and distortion in any subsequent part of the amplification chain. This additional distortion makes active pickups popular among metal and rock guitarists.

The main disadvantages of an active pickup systems are that the system requires a battery power source to operate the active circuitry. Batteries limit the circuit design and functionality, in addition to being an **inconvenience to the musician.** The string instrument with active pickups may contain audio filters, which reduce the dynamic range and mildly distort certain ranges. High output active pickup systems also have an effect on an amplifiers input circuit. This is all to taste. However, when comparing circuitry with an oscilloscope or signal analyser, every aspect of a buffer or preamplifier circuit can be improved by a design that incorporates a rail to rail dual voltage supply preamplifier as found in most amplifiers. These can be designed with the same desired gain and filtration applied.

5.2.1.1.2.6 Stereo and Multiple Pickups with Individual Outputs[17]

This is not a common type of the pickup which is used by Sitar player, but some other string instrument players like guitar players are using this.

Rickenbacker was the first manufacturer who began producing stereo bass guitars with a stereo output for each pickup section. **The neck pickup had one output and the bridge pickup had one.** Also Teisco produced a guitar with a stereo option. Teisco divided the two sections in the upper three strings and the lower three strings for each individual output. The Gittler guitar was an experimental guitar with six pickups, one for each string. The Go Team has modified a Fender Telecaster with an additional rotated pickup for the upper string, causing a simulation of a one string bass sound.

[17] https://en.wikipedia.org/wiki/Pickup_(music_technology)

Gibson also created the HD.6X Pro guitar with The Hex Pickup that captures a separate signal for each individual string and sends it to the on board analog / digital converter, which uses Gibson's digital transport technology to send the signal out of the guitar via Cat 5 Ethernet Cable. The output can be routed as a single summed mono signal to an amplifier or recording console. It can also send the E, A, and D strings to one amp or recording channel and the G, B, and high E to a separate amp or channel. Or it can send the output of all six individual strings to six different amps or channels. These six individualized outputs can used to create various effects.

5.2.1.1.2.7 Merits of Using Pickup over Microphone for String Instruments

- In any kind of the microphone, after generation the sound wave first passes through the air and then enters the mic and processes further. At this stage some outside interference or unwanted noise also enters in to mic along with desired sound (which is known as the **Cross Talk**) and also processes further. This phenomena is totally absent in case of the pickup.
- When we use mic as a transducer for amplification of Sitar sound, its placement position is important because of the property of the mic called 'directivity'. After putting it on proper position Sitarist has to be in that position, while in case of pick up once it is fixed Sitarist can take **any position** of sitting and play Sitar without worrying about sitting position.
- Use of pickup offers better feedback and hence the manageability of the quality of the sound.
- More over to that sound reproduced at the other end using pickup transducer is capable of retaining original tonal quality of the Sitar because of the capacity of covering the desired range of sound frequency.

Thus the use of the pickup gives better feedback and avoids cross talk, convenient during its use, covers whole desired frequency range of the string instruments, generating natural tone at the output end. So it is preferred to use pickup than microphone as a transducer for amplification purpose for the sound of string instruments.

As per me AKG C11 is one of the best quality pick up for Sitar. Technical details of it is as under.

5.2.1.1.2.8 Details of Pick up AKG C411[18]

C411 (L, PP)

HIGH-PERFORMANCE MINIATURE CONDENSER VIBRATION PICKUP

FEEL THE VIBE

For acoustic guitar and other string instruments on stage and in the studio

The C411 is a miniature vibration pickup for acoustic guitar, mandolin, violin and most other string instruments. Its integrated condenser capsule will reproduce the sound of the instrument clearly and without coloration. The C411 ultralight pickup (18g/0.6oz) can be easily attached on or near the bridge with the included nonmarking, solvent-free adhesive pad without changing the balance of the instrument. The C411 is available in two versions. The C411 PP features a MPAV standard XLR connector

[18] http://www.akg.com/pro/p/c411group

while the C411 L provides a professional three-pin mini XLR connector that fits the body-pack transmitters of all AKG wireless microphone systems.

Condenser transducer in sealed enclosure

for clear and uncoloured sound

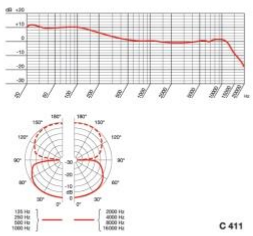
Ultralight vibration pickup

does not change the balance of the instrument

EXPLORE MORE

HARMAN ENTERTAINMENT AND ENTERPRISE SOLUTIONS

TECHNICAL SPECIFICATION

General	Audio frequency bandwidth	10 to 18000 Hz
	Sensitivity	2 mV/Pa
	Electrical impedance	200 Ohms
	Recommended load impedance	1000 Ohms
Polar Patterns	Figure of eight	

Powering Interface	Voltage	9 to 52 V
	Current	2 mA
Audio Output	Type	Mini XLR
	Gender	Female
	Contacts	3-pin
Design	Body	plastic
	Finish	matte black
Dimensions / Weight	Length	9.7 mm
	Width	14.2 mm
	Height	26.7 mm
	Net Weight	18 g

Fig.5.13 Chart of Technical Details of AKG C411

AKG C411 L Microphone Review

By Harmon

Musicians and bands who have cut tracks know how difficult it can be to get just the right sound. Nowhere is this problem more apparent, though, than with string instruments. Fingerpickers in particular find it very hard to record the complex tones and notes that they create. And the more intricate the strings (violin, mandolin, etc.), the tougher it is to reproduce it faithfully through a mic. The AKG C411 L, however, attempts to make recording string instruments much easier.

The first thing you'll notice about the **C411 L** is that it is very small. This allows it to be placed very close to the bridge of any string instrument you're using so that it can pick up even the smallest note variations. The amount of rich tones you can get in this little condenser mic may even astonish seasoned recording veterans. The output is also very loud and earthy, helping the player to be sure that his or her **noodling will come across exactly as it was meant to be heard.**

The best part, though, is that the device clips onto the instrument, meaning that you won't have to drill holes into your pretty acoustic instrument. All in all, the C411 L is a very handy mic, though it does come with a couple drawbacks. For one, it requires phantom power, which means you'll need to buy a cable if you don't already have one. These cables can be a bit pricey, especially after you add in the cost of the mic. Also, the virtuoso players among us might notice that the mic doesn't pick up each and every note if the playing surpasses a certain level. For the vast majority of us, this problem doesn't apply. To the very few players who can play at blistering speeds, though, this might pose a problem. But players of that caliber will likely have better equipment to record with anyway.

THE GOOD

The AKG C411 L is tailor-made for string instruments, picks up finger picking easily and faithfully, and doesn't require you to drill holes in your instruments. It also has suitably loud output capabilities.

THE BAD

It does need an external power source, and it loses its integrity when playing speed gets too high.

KG Pro Audio C411/PP Vibration Pickup for Stringed Instruments

5.0 out of 5 stars

5 customer reviews | 4 answered questions

Condenser transducer in sealed enclosure for clear and uncoloured sound

Ultra-light vibration pickup does not change the balance of the instrument

Figure 8 Polar Pattern accurately captures audio signal from the left and right

Features a MPAV standard XLR connector

About this product

- Description

A precision pickup for guitar, mandolin, and violin, the AKG C 411 L microphone provides clear, pristine sound for the musician. Experience professional-quality sound transfer with this microphone's integrated **condenser capsule technology.** The device attaches to the instrument near the bridge with a non-marking, solvent-free adhesive pad that does nothing to interfere with the function or sound of the instrument itself. With a professional three-pin mini XLR connector, the unit connects with all other AKG body-pack wireless microphone systems for versatility on stage or in the studio. This device features a matte black finish for easy blending.

5.2.2 Changeable Tonal Quality:

Now refer the block diagram-fig. 5.1, after passing through pick up, the signal is passed through the impedance matcher unit which matches the output impedance with the input impedance of the signal processor. **It is of the high importance to match pick up make and the signal processor unit to optimise the best quality of the output.**

To do so is a high precision technical matter.

With help of the attenuator circuit and filters of active band frequency**,** one can change the tonal quality or retain its tonal quality depending upon the wish of an artist. Sound of Sitar in middle octave is ranging from 240 hz to 480 Hz (If tuned to 'C'). For Higher it is double, and half for the lower octave. Depending upon the kind of tone require we can open or close the filter of frequency and hence change the tonal quality. Here we can notice that **an artist can generate required taste of tonal quality without keeping dependency on the audio operator.**

5.2.3. Changeability of Sustention of the Sound.

It is noticed in the previous chapter that an artist may require different sustention of the sound during his single program. He has one option of changing the Sitar with different length or do something electronically. The signal processor contains various filters, by adjusting them a required sustention of sound can be increased or decreased. Some good quality processors have more than two three band pass to define sustention time more precisely. Processor also possesses the knobs for course tuning and fine tuning for the precise control of output.

5.2.4 Tuning

Now feature of tuning apparatus for Sitar will be fulfilled. For this please refer to the block diagram 5.1. The point from where output signal goes to the amplifier, it is also fetched to the Electronic tuner (tuning device). So basically it is a sample of an output. **Electronic Tuner** is a device that detects and displays the pitch of musical notes played on a musical instrument.

'Pitch' is the highness or lowness of a musical note, which is typically measured in Hertz. Simple tuners indicate typically with an analog needle-dial, LEDs, or an LCD screen. Whether a pitch is lower, higher, or equal to the desired pitch. In the recent years, apps. are also available to convert a smartphone, tablet, or personal computer into a tuner.

The simplest tuners detect and display tuning only for a single pitch often 'A' or 'E'- or for a small number of pitches, such as the six used in the standard tuning of a guitar (E,A,D,G,B,E). More complex tuners offer chromatic tuning for all 12 pitches of the equally tempered octave. Some electronic tuners offer additional features, such as pitch calibration, temperament options, the sounding of a desired pitch through an amplifier plus speaker, and adjustable 'read-time' settings that affect how long the tuner takes to measure the pitch of the note.

Regular Types

Regular electronic tuners contain either an input jack for electric instruments (usually a 1/4" patch cord input), a microphone, or a clip-on sensor (e.g., a **piezoelectric pickup)** or some combination of these inputs. Pitch detection circuitry drives some type of display (an analog needle, an LCD simulated image of a needle, LED lights, or a spinning translucent disk illuminated by a strobing backlight). Some tuners have an output, or through-put, so the tuner can connect 'in-line' from an electric instrument to an instrument amplifier or mixing console. Small tuners are usually battery powered. Many battery powered tuners also have a jack for an optional AC power supply.

Fig. 5.14 Clip on Type Tuner.

Most musical instruments generate a fairly complex waveform. It contains a number of harmonic partials, including the fundamental frequency (which a typical listener perceives as the pitch of the note) and additional ‘harmonics’ (also called ‘partials’ or ‘overtones). Each instrument produces different ratios of harmonics, which is what makes notes of the same pitch played on different instruments (e.g., an A 440Hz note played on oboe, violin or electric guitar) sound different. As well, this waveform constantly changes. This means that for non-strobe tuners to be accurate, the **tuner must process a number of cycles and use the pitch average to drive its display**. Background noise from other musicians or harmonic overtones from the musical instrument can impede the electronic tuner from ‘locking’ onto the input frequency. This is why the needle or display on regular electronic tuners tends to waver when a pitch is played. Small movements of the needle, or LED, usually represent a tuning error of 1 cent. Typical accuracy of these types of tuners is around +/- 3 cents. Some inexpensive LED tuners may drift by as much as +/- 9 cents.

'**Clip-on**' tuners typically attach to instruments with a spring-loaded clip that has a built-in contact microphone. Clipped onto a Sitar peg, it senses the vibrations from the instrument, even in a noisy environment.

The least expensive models only detect and display a small number of pitches, often those pitches that are required to tune a given instrument (e.g., E, A, D, G, B, E of standard guitar tuning). While this type of tuner is useful for bands that only use stringed instruments such as guitar and electric bass, it is not that useful for tuning brass or woodwind instruments. Tuners at the next price point offer chromatic tuning, the ability to detect and assess all the pitches in the chromatic scale (e.g., C, C♯, D, D♯, etc.). Chromatic tuners can be used for B♭ and E♭ scaled brass instruments, such as saxophones and horns. Many models have circuitry that automatically detects which pitch is being played, and then compares it against the correct pitch. Less expensive models require the musician to specify the target pitch via a switch or slider. Most low- and mid-priced electronic tuners only allow tuning to an equal temperament scale.

Many models let the user select reference pitches other than A440. On many electronic tuners, the user can select a different note.

5.2.4.1 Application in Classical Music

In classical music, there is a long-standing tradition to **tune 'by ear'**, by adjusting the pitch of instruments to a reference pitch.

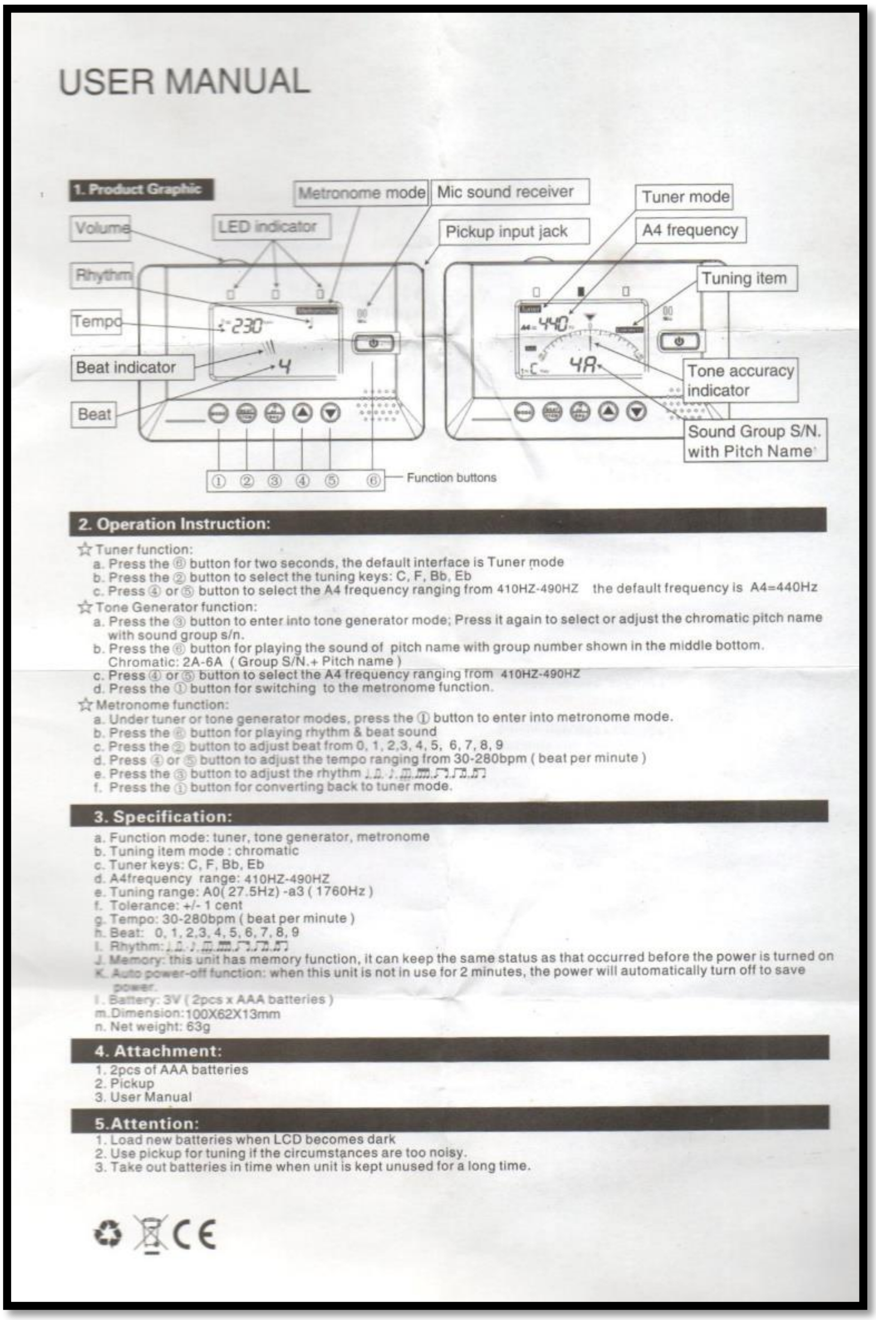
In an orchestra, the oboe player gives a 440 Hz '**A**', and the different instrument sections tune to this note.

In chamber music, either one of the woodwind player gives an 'A', or if none is present, one of the string players, usually the first violinist, bows his open 'A' string. If an orchestra is accompanying a piano concerto, the first oboist takes the 'A' from the piano and then plays this pitch for the rest of the orchestra (a somewhat unscientific procedure, but consecrated by longstanding usage). **Despite this tradition of tuning by ear, electronic tuners are still widely used in classical music.**

Classical performers also use tuners off-stage for practice purposes or to check their tuning (or, with the further aid of a speaker, to practice ear training).In general to tune the Sitar requires peaceful environment (even fan noise can influence the tuning process.) In the orchestra, environment is not suitable for tuning the Sitar. And situation worsens when show is on and tuning is to be done. Tuner unit if made an inbuilt part of the Sitar, facilitate a player to tune Sitar in more precise and scientific manner and tune all rest instruments or group of the Sitar at standardized level. **At this time electronic tuner enables Sitarist to tune the Sitar during live show making Sitar a competent musical instrument in contemporary music –orchestra with Indian and other western instruments.**

Following is the technical detail of some different tuners[19]

[19] http://www.korg.com/in/products/tuners/ca_40/



USER MANUAL

1. Product Graphic

2. Operation Instruction:

☆ Tuner function:
a. Press the ⑥ button for two seconds, the default interface is Tuner mode
b. Press the ② button to select the tuning keys: C, F, Bb, Eb
c. Press ④ or ⑤ button to select the A4 frequency ranging from 410HZ-490HZ the default frequency is A4=440Hz

☆ Tone Generator function:
a. Press the ③ button to enter into tone generator mode; Press it again to select or adjust the chromatic pitch name with sound group s/n.
b. Press the ⑥ button for playing the sound of pitch name with group number shown in the middle bottom. Chromatic: 2A-6A (Group S/N.+ Pitch name)
c. Press ④ or ⑤ button to select the A4 frequency ranging from 410HZ-490HZ
d. Press the ① button for switching to the metronome function.

☆ Metronome function:
a. Under tuner or tone generator modes, press the ① button to enter into metronome mode.
b. Press the ⑥ button for playing rhythm & beat sound
c. Press the ② button to adjust beat from 0, 1, 2,3, 4, 5, 6, 7, 8, 9
d. Press ④ or ⑤ button to adjust the tempo ranging from 30-280bpm (beat per minute)
e. Press the ③ button to adjust the rhythm
f. Press the ① button for converting back to tuner mode.

3. Specification:

a. Function mode: tuner, tone generator, metronome
b. Tuning item mode : chromatic
c. Tuner keys: C, F, Bb, Eb
d. A4frequency range: 410HZ-490HZ
e. Tuning range: A0(27.5Hz) -a3 (1760Hz)
f. Tolerance: +/- 1 cent
g. Tempo: 30-280bpm (beat per minute)
h. Beat: 0, 1, 2,3, 4, 5, 6, 7, 8, 9
I. Rhythm:
J. Memory: this unit has memory function, it can keep the same status as that occurred before the power is turned on
K. Auto power-off function: when this unit is not in use for 2 minutes, the power will automatically turn off to save power.
l. Battery: 3V (2pcs x AAA batteries)
m. Dimension: 100X62X13mm
n. Net weight: 63g

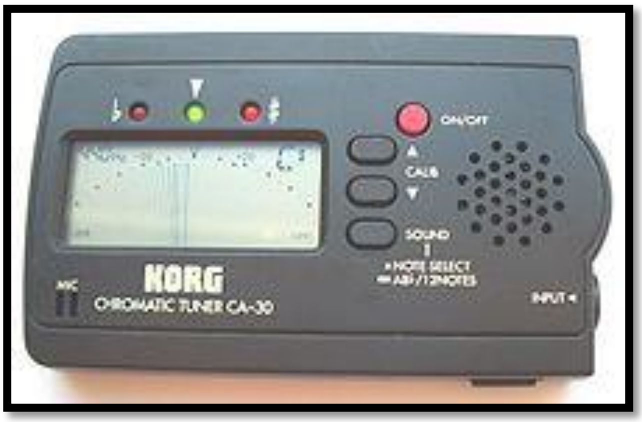
4. Attachment:

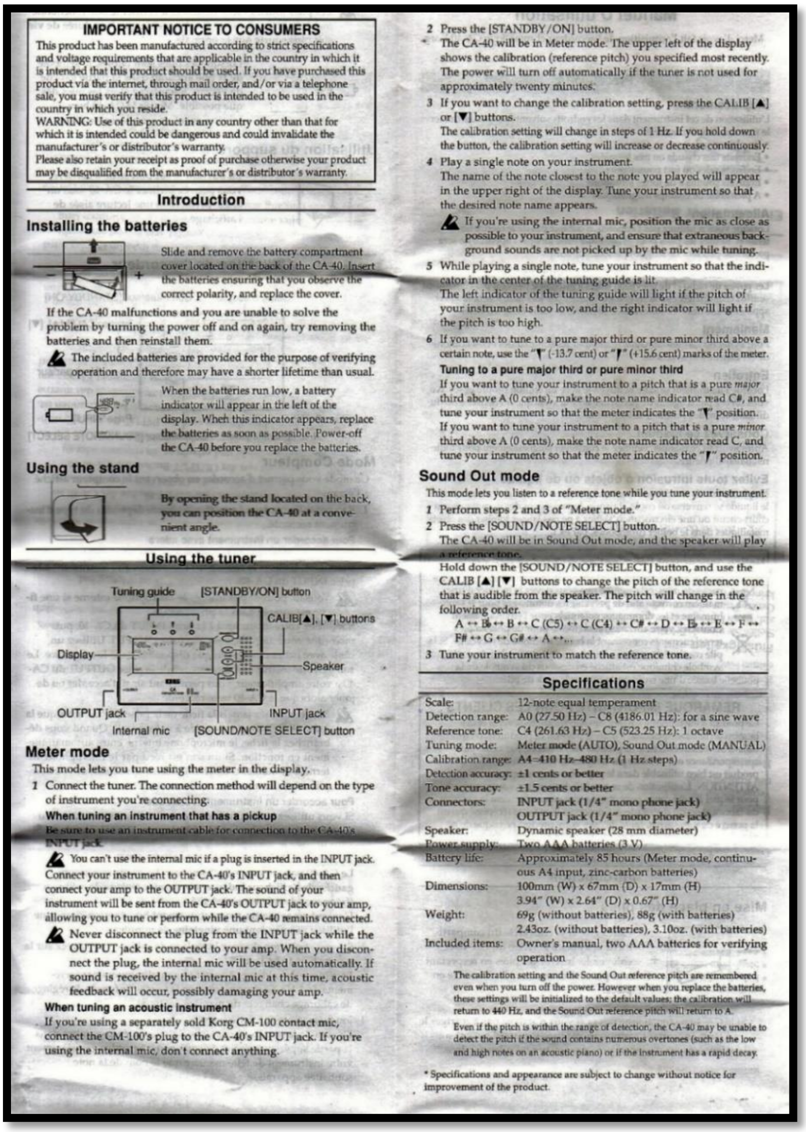
1. 2pcs of AAA batteries
2. Pickup
3. User Manual

5. Attention:

1. Load new batteries when LCD becomes dark
2. Use pickup for tuning if the circumstances are too noisy.
3. Take out batteries in time when unit is kept unused for a long time.

Fig 5.15 Tuner SX 60 MT with Technical Detail





IMPORTANT NOTICE TO CONSUMERS

This product has been manufactured according to strict specifications and voltage requirements that are applicable in the country in which it is intended that this product should be used. If you have purchased this product via the internet, through mail order, and/or via a telephone sale, you must verify that this product is intended to be used in the country in which you reside.
WARNING: Use of this product in any country other than that for which it is intended could be dangerous and could invalidate the manufacturer's or distributor's warranty.
Please also retain your receipt as proof of purchase otherwise your product may be disqualified from the manufacturer's or distributor's warranty.

Introduction

Installing the batteries

Slide and remove the battery compartment cover located on the back of the CA-40. Insert the batteries ensuring that you observe the correct polarity, and replace the cover.

If the CA-40 malfunctions and you are unable to solve the problem by turning the power off and on again, try removing the batteries and then reinstall them.

The included batteries are provided for the purpose of verifying operation and therefore may have a shorter lifetime than usual.

When the batteries run low, a battery indicator will appear in the left of the display. When this indicator appears, replace the batteries as soon as possible. Power-off the CA-40 before you replace the batteries.

Using the stand

By opening the stand located on the back, you can position the CA-40 at a convenient angle.

Using the tuner

Tuning guide
[STANDBY/ON] button
CALIB[▲], [▼] buttons
Display
Speaker
OUTPUT jack
INPUT jack
Internal mic
[SOUND/NOTE SELECT] button

Meter mode

This mode lets you tune using the meter in the display.

1 Connect the tuner. The connection method will depend on the type of instrument you're connecting.

When tuning an instrument that has a pickup

Be sure to use an instrument cable for connection to the CA-40's INPUT jack.

You can't use the internal mic if a plug is inserted in the INPUT jack.

Connect your instrument to the CA-40's INPUT jack, and then connect your amp to the OUTPUT jack. The sound of your instrument will be sent from the CA-40's OUTPUT jack to your amp, allowing you to tune or perform while the CA-40 remains connected.

Never disconnect the plug from the INPUT jack while the OUTPUT jack is connected to your amp. When you disconnect the plug, the internal mic will be used automatically. If sound is received by the internal mic at this time, acoustic feedback will occur, possibly damaging your amp.

When tuning an acoustic instrument

If you're using a separately sold Korg CM-100 contact mic, connect the CM-100's plug to the CA-40's INPUT jack. If you're using the internal mic, don't connect anything.

2 Press the [STANDBY/ON] button.
The CA-40 will be in Meter mode. The upper left of the display shows the calibration (reference pitch) you specified most recently. The power will turn off automatically if the tuner is not used for approximately twenty minutes.

3 If you want to change the calibration setting, press the CALIB [▲] or [▼] buttons.
The calibration setting will change in steps of 1 Hz. If you hold down the button, the calibration setting will increase or decrease continuously.

4 Play a single note on your instrument.
The name of the note closest to the note you played will appear in the upper right of the display. Tune your instrument so that the desired note name appears.

If you're using the internal mic, position the mic as close as possible to your instrument, and ensure that extraneous background sounds are not picked up by the mic while tuning.

5 While playing a single note, tune your instrument so that the indicator in the center of the tuning guide is lit.
The left indicator of the tuning guide will light if the pitch of your instrument is too low, and the right indicator will light if the pitch is too high.

6 If you want to tune to a pure major third or pure minor third above a certain note, use the "▼" (-13.7 cent) or "▼" (+15.6 cent) marks of the meter.

Tuning to a pure major third or pure minor third

If you want to tune your instrument to a pitch that is a pure *major* third above A (0 cents), make the note name indicator read C#, and tune your instrument so that the meter indicates the "▼" position.
If you want to tune your instrument to a pitch that is a pure *minor* third above A (0 cents), make the note name indicator read C, and tune your instrument so that the meter indicates the "▼" position.

Sound Out mode

This mode lets you listen to a reference tone while you tune your instrument.

1 Perform steps 2 and 3 of "Meter mode."

2 Press the [SOUND/NOTE SELECT] button.
The CA-40 will be in Sound Out mode, and the speaker will play a reference tone.
Hold down the [SOUND/NOTE SELECT] button, and use the CALIB [▲] [▼] buttons to change the pitch of the reference tone that is audible from the speaker. The pitch will change in the following order.
A ↔ B♭ ↔ B ↔ C (C5) ↔ C (C4) ↔ C# ↔ D ↔ E♭ ↔ E ↔ F ↔ F# ↔ G ↔ G# ↔ A ↔...

3 Tune your instrument to match the reference tone.

Specifications

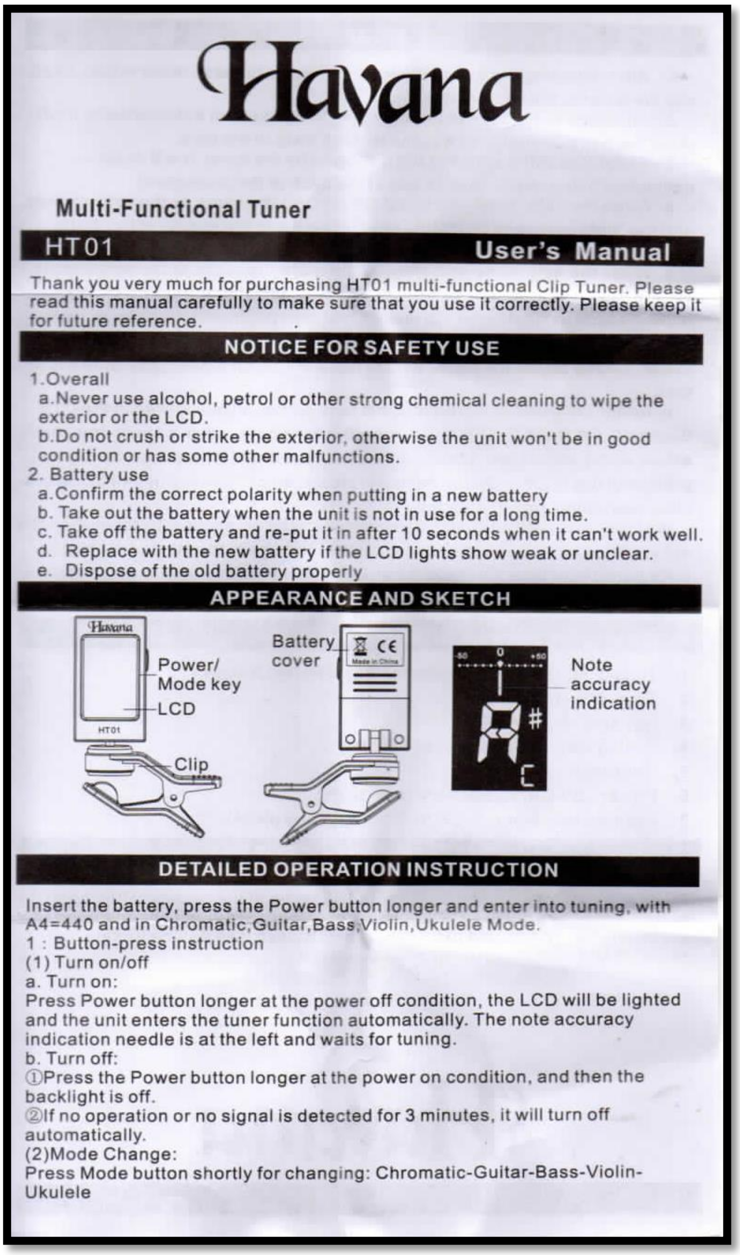
Scale:	12-note equal temperament
Detection range:	A0 (27.50 Hz) – C8 (4186.01 Hz): for a sine wave
Reference tone:	C4 (261.63 Hz) – C5 (523.25 Hz): 1 octave
Tuning mode:	Meter mode (AUTO), Sound Out mode (MANUAL)
Calibration range:	A4=410 Hz–480 Hz (1 Hz steps)
Detection accuracy:	±1 cents or better
Tone accuracy:	±1.5 cents or better
Connectors:	INPUT jack (1/4" mono phone jack) OUTPUT jack (1/4" mono phone jack)
Speaker:	Dynamic speaker (28 mm diameter)
Power supply:	Two AAA batteries (3 V)
Battery life:	Approximately 85 hours (Meter mode, continuous A4 input, zinc-carbon batteries)
Dimensions:	100mm (W) x 67mm (D) x 17mm (H) 3.94" (W) x 2.64" (D) x 0.67" (H)
Weight:	69g (without batteries), 88g (with batteries) 2.43oz. (without batteries), 3.10oz. (with batteries)
Included items:	Owner's manual, two AAA batteries for verifying operation

The calibration setting and the Sound Out reference pitch are remembered even when you turn off the power. However when you replace the batteries, these settings will be initialized to the default values; the calibration will return to 440 Hz, and the Sound Out reference pitch will return to A.

Even if the pitch is within the range of detection, the CA-40 may be unable to detect the pitch if the sound contains numerous overtones (such as the low and high notes on an acoustic piano) or if the instrument has a rapid decay.

* Specifications and appearance are subject to change without notice for improvement of the product.

Fig. 5.16 Pocket-Sized Korg chromatic LCD tuner, with Simulated Analog Indicator Needle and Technical Details



Havana

Multi-Functional Tuner

HT01 User's Manual

Thank you very much for purchasing HT01 multi-functional Clip Tuner. Please read this manual carefully to make sure that you use it correctly. Please keep it for future reference.

NOTICE FOR SAFETY USE

1.Overall
a.Never use alcohol, petrol or other strong chemical cleaning to wipe the exterior or the LCD.
b.Do not crush or strike the exterior, otherwise the unit won't be in good condition or has some other malfunctions.
2. Battery use
a.Confirm the correct polarity when putting in a new battery
b. Take out the battery when the unit is not in use for a long time.
c. Take off the battery and re-put it in after 10 seconds when it can't work well.
d. Replace with the new battery if the LCD lights show weak or unclear.
e. Dispose of the old battery properly

APPEARANCE AND SKETCH

Power/ Mode key
LCD
Clip
Battery cover
Note accuracy indication

DETAILED OPERATION INSTRUCTION

Insert the battery, press the Power button longer and enter into tuning, with A4=440 and in Chromatic,Guitar,Bass,Violin,Ukulele Mode.
1 : Button-press instruction
(1) Turn on/off
a. Turn on:
Press Power button longer at the power off condition, the LCD will be lighted and the unit enters the tuner function automatically. The note accuracy indication needle is at the left and waits for tuning.
b. Turn off:
①Press the Power button longer at the power on condition, and then the backlight is off.
②If no operation or no signal is detected for 3 minutes, it will turn off automatically.
(2)Mode Change:
Press Mode button shortly for changing: Chromatic-Guitar-Bass-Violin-Ukulele

Fig. 5.17 Havana Make Clip on Tuner with Technical Manual

5.2.5 Recording of Sound

5.2.6 Retrieval of Sound

A single unit is available giving both the facilities i.e. recording and replaying the recorded sound.

The junction from where output is fetched to the amplifier, a sample is taken and sent for recording into sound recorder. And if recorded sound is replayed it is connected at input point as shown in the block diagram fig. 5.1, at switch point S1.

Both the features are achieved with a single unit. This unit is capable of recording about 24 hrs. X 100 days. (Or 46 days x 24 hrs. depending on its memory capacity)This unit records data in its EPROM unit and free of hard disk storage system. Can be easily attached to computer system also.

Illustration is given for Sony sound recorder and retrieval unit[20]

5.2.6.1 Details of SONY Sound Recorder/ Retrieval Unit

Details of Sony sound recorder is as below.

[20] https://www.sony.com/electronics/voice-recorders/icd-px333-series/specifications#features

SONY

MONO DIGITAL VOICE RECORDER WITH PC LINK

FULL SPECIFICATIONS AND FEATURES

Overall	BUILT-IN MEMORY 4 GB PC CONNECTIVITY YES BUILT-IN MICROPHONE Monaural RECORDING FORMAT MP3 PLAYBACK FORMAT MP3	BATTERY TYPE AAA x 2 MAXIMUM NUMBER OF FOLDERS 400 MAXIMUM NUMBER OF FILES 4074 MAXIMUM NUMBER OF FILES IN ONE FOLDER 199
Inputs and Outputs	HEADPHONE JACK YES MICROPHONE IN YES USB PORT YES	DC IN JACK NO MEMORY CARD SLOT "M2"/microSD

Recording Features	SCENE SELECT YES FM RADIO RECORDING NO AM RADIO RECORDING NO MANUAL RECORDING LEVEL ADJUSTMENT NO LOW-CUT FILTER YES LIMITER FUNCTION NO	ADD/OVERWRITE RECORDING YES PRE-RECORDING NO CROSS-MEMORY RECORDING NO VOR YES SYNC RECORDING FUNCTION NO RECORDING MONITOR YES
Maximum Recording time	MP3 8 KBPS (MONAURAL) 1073 Hrs 0 Min MP3 48 KBPS (MONAURAL) 178 Hrs 0 Min MP3 128 KBPS 67 Hrs 5 Min MP3 192 KBPS 44 Hrs 40 Min	MP3 320 KBPS - LPCM 44.1 KHZ, 16 BIT - LPCM 96.1 KHZ, 24 BIT -

Battery Life for recording	HVXC 2 KBPS (MONAURAL) - MP3 8 KBPS (MONAURAL) 96 Hrs 0 Min MP3 48 KBPS (MONAURAL) 72 Hrs 0 Min MP3 128 KBPS 65 Hrs 0 Min	MP3 192 KB PS - MP3 320 KBPS - LPCM 44.1 KHZ, 16 BIT - LPCM 96.1 KHZ, 24 BIT -
Frequency response	HVXC 2 KBPS (MONAURAL) - MP3 8 KBPS (MONAURAL) 75 - 3500 Hz MP3 48 KBPS (MONAURAL) 75 - 14,000 Hz MP3 128 KBPS 75 - 17,000 Hz	MP3 192 KBPS 75 - 20,000 Hz MP3 320 KBPS - LPCM 44.1 KHZ, 16 BIT - LPCM 96.1 KHZ, 24 BIT -

Playback Features	DIGITAL NOISE CANCELING NO FULL DIGITAL AMPLIFIER NO DIGITAL PITCH CONTROL (SPEED CONTROL) YES KEY CONTROL NO NOISE CUT YES (Intelligent Noise Cut)	DIGITAL VOICE UP NO GRAPHIC EQUALIZER NO A-B REPEAT YES EASY SEARCH YES ALARM PLAYBACK YES
Edit Features	TRACK MARK YES ERASE YES PROTECT YES DIVIDE YES	MOVE FILE YES FILE COPY YES FOLDER NAME CHANGE (WITHOUT USING PC) NO

Additional Features	CALENDAR SEARCH NO LCD BACKLIGHT NO	USB CONNECTION CHARGING NO
Speaker	SPEAKER POWER OUTPUT 300 mW	SPEAKER SIZE 1.10 in
Others	BATTERY TYPE (PROVIDED) Dry Battery (Alkaline, Size AAA) DIMENSIONS (INCH, W/H/D) 1.48 x 4.49 x 0.82	WEIGHT (OZ, INCLUDING BATTERIES) 2.61
Menu	MENU LANGUAGE English, Spanish, French, German, Italian, Russian	

Supplied Software & Supplied Accessories	PC APPLICATION SOFTWARE (SOUND ORGANIZER) YES PC APPLICATION SOFTWARE (DRAGON NA TURALLY SPEAKING/DVD-ROM) - STEREO HEADPHONES - STAND - CARRYING POUCH/CASE - RECHARGEABLE BATTERY (NIMH, SIZE AAA) -	DRY BATTERY (ALKALINE, SIZE AAA) Yes (x2) STEREO MICROPHONE WITH SHIRT-CLIP - USB CABLE YES USB CONNECTION SUPPORT CABLE - HANDSTRAP -
Package	PACKAGING LANGUAGE English, French	PACKAGE STYLE Paper Box with Hook
Quick Start Guide	ENGLISH YES SPANISH YES	FRENCH YE S

Fig. 5.18 Table of Technical Details of Sony Sound Recorder

5.2.6.2 Zoom Sound Recorder / Retrieval Unit

Detail of another sound recorder with SD card is as follows:

ZOOM H1 PORTABLE DIGITAL AUDIO RECORDER[21]

A compact and easy to use product that keeps the users focused on their creativity and the task at hand. One of the most noticeable feature about the zoom H1 is its size. Its candy-bar design is ultra-portable and can easily fit in your pocket or kit-bag. Whether you're recording notes for class, making a podcast, or even film-making, Zoom H1 has you covered, Simplicity is one of the most defining feature of the Zoom H1. Stereo X-Y mics, easy to read display panel lets you read the sound level, recording time, battery info etc. A single button start-stop feature is inspired by the old-school analog form and is really a great way to keep things simple. For musicians the Zoom h1 is surely going to be a great tool. From band practice to recording riffs, the Zoom H1 does it all.

X/Y recording made simple

The X/Y technique provides a great way to cover a wide area while still capturing sound sources in the center with clarity and definition, making it perfect for all types of live stereo recording.
The H1's built-in X/Y microphone provides two matched unidirectional microphones set at a 90 degree angle relative to one another, optimum for most stereo recording applications.
Alternatively, you can connect a pair of external microphones or line level signal to the H1's Mic/Line Input mini phone jack for X/Y or other types of stereo recording.

[21] https://www.zoom-na.com/products/field-video-recording/field-recording/zoom-h1-handy-recorder

The Ins and Outs

The H1 Mic/Line Input is a stereo ⅛" mini phone jack that can accept two mic- and/or line-level signals. Condenser microphones requiring Plug-In Power (2.5 volts) can be connected to this jack.

The H1 Line/Headphones Output is a stereo ⅛" phone jack with a dedicated volume control. Headphones can be connected here for private monitoring. There's also a built-in speaker on the back panel for fast monophonic monitoring of the recorded signal without the need to make any connections.

The H1's USB port provides a digital output of the stereo mix and allows data to be sent to and from your computer. From there, it can be imported into editing software such as the supplied Wave Lab LE. It also allows the H1 to be used as a 2-in/2-out audio interface and USB microphone, as well as a micro SD card reader.

Auto Level and low cut filter

The H1's Auto Level function sets input gain automatically to prevent overload and distortion. Alternatively, input level can be set manually. The H1 also provides a built-in low cut filter for the elimination of pops, wind noise, blowing, and other kinds of low frequency rumble.

File types supported

The H1 Mic/Line Input is a stereo ⅛" mini phone jack that can accept two mic- and/or line-level signals. Condenser microphones requiring Plug-In Power (2.5 volts) can be connected to this jack.
The H1 Line/Headphones Output is a stereo ⅛" phone jack with a dedicated volume control. Headphones can be connected here for private monitoring. There's also a built-in speaker on the back panel for fast monophonic monitoring of the recorded signal without the need to make any connections.
The H1's USB port provides a digital output of the stereo mix and allows data to be sent to and from your computer. From there, it can be imported into editing software such as the supplied Wave Lab LE.

Battery life and recording times

The H1 requires just a single AA battery—either alkaline or rechargeable NiMH—for power. Battery life when using an alkaline battery is up to 10 hours, even during continuous recording. Alternatively, you can use an optional AD-17 AC adapter, which allows you to power the H1 from any standard wall socket.

The H1 records directly to micro SD and micro SDHC cards, up to 32 gigabytes. The chart below shows maximum recording time with different file formats and card sizes.

REC Format		Micro SD/SDHC Card Capacity				
		2GB	4GB	8GB	16GB	32GB
MP3	128kbps	34hrs43min	69hrs26min	138hrs53min	277hrs46min	555hrs33min
	256kbps	17hrs21min	34hrs43min	69hrs26min	138hrs53min	277hrs46min
	320kbps	13hrs53min	27hrs46min	55hrs33min	111hrs06min	222hrs12min
WAV	16bit / 44.1kHz	3hrs08min	6hrs17min	12hrs35min	25hrs11min	50hrs23min
	24bit / 48kHz	1hrs55min	3hrs51min	7hrs42min	15hrs25min	30hrs51min
	24bit / 96kHz	57min	1hrs55min	3hrs51min	7hrs42min	15hrs25min

Features:

- Built-in 90˚ X/Y stereo mic.
- Stereo ⅛" Mic/Line Input mini phone jack with Plug-in power (2.5V).
- Stereo ⅛" Phones/Line Output jack with dedicated volume control.
- Built-in reference speaker for fast monitoring.

- Backlit LCD display.
- Records directly to micro SD and micro SDHC cards up to 32 GB.
- Supports up to 24-bit/96 kHz audio in BWF-compliant WAV or a variety of MP3 formats.
- Auto Level for automatic control of input level.
- Low-cut filter for elimination of wind noise and rumble.
- Up to 99 marks per recording.
- USB port for data transfer to computer and use as an audio interface and USB microphone.
- SD card reader function.
- Mounts directly to tripod, or to mic stand or DSLR with optional adapter.
- Runs on only 1 standard AA alkaline or NiMH rechargeable battery.
- Up to 10 hours of operation with a single AA alkaline battery.

SPECS:

- Simultaneous recording tracks: 2
- Simultaneous playback tracks : 2
- Functions: Lo-cut Filter, Auto REC Level, Marker
- Recording/playback format: WAV: 44.1 / 48 / 96kHz, 16- / 24-bit
- MP3: 44.1kHz 48/56/64/80/96/112/128/160/192/224/256/320kbps
- A/D conversion: 24-bit, 128x oversampling
- D/A conversion: 24-bit, 128x oversampling
- Signal processing: 32-bit
- Recording media: microSD card (16MB - 2GB)
- microSDHC card (4GB - 32GB)
- Display: 127 segment custom LCD (with backlight)
- Built-in stereo mic: Unidirectional condenser
- Gain: 0 to +39dB
- Minimum gain with digital attenuation: -28dB
- Maximum sound pressure level: 120dB SPL
- Mic/line input: 1/8" stereo phone jack (Plug-in power supported)
- Input Impedance: 2kΩ (Input level: 0 to -39dBm)
- Phones/line output: 1/8" stereo phone jack
- Output load impedance: 10kΩ or more
- Rated output level: -10dBm
- Phones output level: 20mW + 20mW into 32Ω load
- Output load impedance: 10kΩ or more
- Rated output level: -10dBm
- USB interface: Type: Mini-B type (USB 2.0 High Speed compatible), Mass Storage Class operation
- Format: 44.1 kHz/16-bit or 48 kHz/16-bit
- Power requirements: Alkaline or Ni-MH AA battery x 1, or AC adapter (AD-17, USB to AC type)
- Battery life (alkaline batteries): 10 hours (MP3), 9.5 hours (WAV)
- Dimensions: 44(W) x 136(D) x 31(H)mm
- Weight: 60g (without batteries)

Fig.5.19 Table of technical detail of Zoom Recorder

5.2.7 Monitoring Output of Sound

As per diagram 5.1, a sample of the output is taken from the output junction and it is send to a good quality headphone. Using this an artist will come to know outgoing sound quality. Hence he will be able to do necessary changes in controller unit if required and get desired output. Using the switch combination provided, in block he can listen the outgoing sound of Sitar individually, or combined sound of Sitar with other instruments.

5.2.8 Volume Control:

This feature is an inbuilt part of the signal processor. It is just a regular volume control. By varying this we can increase or decrease the amplitude of the signal sent for an amplification. If signal with more amplitude is send, more amplitude signal will be gained by amplifier resulting more power and hence larger volume. Similarly if signal with less amplitude is sent to amplifier it will amplify a less amplitude signal causing lower volume sound. This application is most frequently used by an artist. Some times because of larger amplification a whistling is seen in the output. At this time an artist himself can control this effect with help of a volume control and / or an attenuator.

With combination of feedback head phone and volume control a perfect sound balancing can be achieved by instrumentalist, keeping himself in dominating position in the show.

5.2.9 Mute

This feature is provided in form of a small button on a panel of a signal processor. If during live show if tuning happen to be done then this feature provides a very sophisticated feature of muting. This is a very small feature but impacting a lot sometimes. This mute key is provided in the series of a path reaching to the input of an amplifier. Pressing it once, the mute function is enabled, and disabled pressing it twice. This function can also be utilised in some other unusual condition happening on the stage.

5.2.10 Pictures of the Designed unit

5.2.10.1 Face Plate of Docstation Indicating the Features

Fig. 5.20 Face Plate of Docstation Indicating the Features

5.2.10.2 Docstation

Fig. 5.21 Docstation

5.2.10.3 Front and Side View of Sitar with Docstation

Fig. 5.22 Front and Side View of Sitar with Docstation

5.2.10.4 Unit in Practical Use

Fig. 5.23 Researcher Taking Trial of the Instrument, Docstation and Sitar at his Studio

Fig. 5.24 Researcher Validating Performance of Equipment with Prominent Artist

Fig. 5.25 Researcher Demonstrating Docstation and Sitar in a Chamber Concert

Fig. 5.26 Prominent Artists with Docstation and Sitar

THE TIMES OF INDIA, AHMEDABAD
THURSDAY, JULY 24, 2014

Vadodara engineer 'electrifies' sitar

Prashant.Rupera
@timesgroup.com

Vadodara: Love for music has made this electronics engineer modify the long-necked Indian classical music instrument – sitar. Bharat Desai, a city-based electronics engineer, who has just completed his masters from M S University's Faculty of Performing Arts (FPA) with first class, has developed a new musical instrument 'Bharatar' that fine tunes the minute 'dhvanis' of sitar with the right kind of amplification.

Interestingly, 'Bharatar' also provides a facility whereby recording of music played from sitar can happen non-stop for 100 days. "I had started working on developing this new instrument since the time I realized that unlike other musical instruments, techniques for production and propagation of music of sitar have not been developed," says Desai, who had also created a fusion of sorts in his career by completing both electronics engineering from MSU's Faculty of Technology and Engineering and diploma in music (sitar) from FPA in 1991.

"Bharatar is a result of fusion of technology and music. I took opinion of several prominent artists to make changes in sitar so that low frequency voice of sitar can be amplified without changing its natural tone," says

Ashwin Rajput

Bharat Desai has developed a new musical instrument 'Bharatar' that fine tunes the minute 'dhvanis' of sitar with right kind of amplification

Desai, owner of a computer hardware firm, who now plans to register for a PhD and file a patent for his development. 'Bharatar' also displays tuning positions of each of the 23 strings of a sitar. "This helps an amateur or even a professional artist to tune the strings despite loud noise of other musical instruments in an orchestra," says Desai.

"When an artist plays a traditional sitar, the sitar produces spiky notes. This instrument improves sustainability of the music produced by the sitar," says Desai, adding that 'Bharatar' also has facility of volume, distortion and fade controls. "This new instrument has helped improve tonal quality of sitar," says Desai's guru and associate professor in instrumental music (sitar) at FPA A S Pathan, who has used 'Bharatar' for nearly six months.

Fig. 5.27 Technical Details of the Researcher's Unit in Times of India (Ahmedabad)


TIMES CITY

NOW, MERGE PLOTS TO BUILD BIGGER HOUSES | 4

DREAMS NOT DISABLED FOR THESE BRAVEHEARTS | 4

TIMES OF INDIA

This electronics engineer 'electrifies' traditional sitar

Fig.5.28 Technical Details of the Researcher's Unit in Times of India (Baroda)


RARE COLLECTION ભરત દેસાઈએ મોડિફાઇડ સિતારને 'ભરતાર' નામ આપ્યું, જેમાં રેકોર્ડિંગ પણ કરી શકાય છે.

આ એન્જિનિયરે સિતારને હાઇટેક બનાવી

પાંચ વર્ષ સતત પ્રયાસના અંતે ભરતભાઈએ સિતાર બનાવી છે

સિતાર બે ભાગમાં વહેંચાયેલી છે

ડોક્સ્ટેશન

ટ્રેડિશનલ યુનિટ

એ.આર.રહેમાનની કોન્સર્ટમાં અસદ ખાને વગાડી હતી

સિતાર આર્ટ અને ટેક્નોલોજીનું ફ્યુઝન છે

Fig.5.29 Technical Details of the Researcher's Unit in Divya Bhaskar (Baroda)