

RF Sounding: Generating Sounds from Radio Frequencies

Claudia Rinaldi, Fabio Graziosi, Luigi Pomante and Francesco Tarquini
*Center of Excellence DEWS, University of L'Aquila
Italy*

1. Introduction

In this chapter we present an innovative way of exploiting technological innovations brought by Wireless Sensor Networks (WSNs) functionalities and cellular communications in two important social fields: art and education. The result is an artistic installation built through the use of innovative technologies that can also be used with educational purposes.

In recent years there has been a growing interest on the theme of multidisciplinary as well as exploitation of new technologies in the artistic field. Artistic installations and performance have exploited the most recent technological innovations, introducing new needs and requirements to be satisfied from a scientific point of view.

On the other hand the problem of the electromagnetic pollution has recently become an important theme of discussion for both scientific and media communities and it has often been faced with incomplete, where not wrong, knowledge of the phenomena involved. This is especially true while concerning with cellular networks where the most popular belief is that transmitting antenna produce the most of the electromagnetic radiation. RF Sounding will provide the user with the possibility of understanding the real behaviour of cellular communications in terms of RFs transmission power.

A classification of interactive art installations, organized according to which software technology (like for example MAX/MSP) and software engineering processes have been used, is presented in (1). (2) presents the software engineering process that has led to the development of *Flyndre*, an installation at the coast of Norway, where the composition is changed according to weather parameters. In this respect, (3) presents the technological and artistic processes around *Sonic Onyx*, an installation in Trondheim where the audience can send Bluetooth files, which are reconstructed to new sound compositions and played.

The usefulness of technology into art expressions and education has also been demonstrated in two projects from MIT. *Musicpainter*, (4), is a networked, graphical composing environment that aims to encourage sharing of music creation and collaboration within the composing process. Indeed a composer could start with a small idea (e.g. only rhythmic or melodic), share it in the network and receive suggestions from other composers. *TablaNet*, (5), is an intelligent system that listens to the audio input at one end and synthesizes a predicted audio output at the other, when live music event where musicians are located remotely from each

other. Other examples of artistic performances that are possible because of technology are the ones presented by the Opera of the Future Group from MIT media lab, (6). For instance "*Death and the Powers*" is a groundbreaking opera that brings a variety of technological, conceptual, and aesthetic innovations to the theatrical world. Musical performances with educational purposes too are also possible by using *Reactable*, (7), an electronic music instrument realized by the Music Technology Group of Pompeu Fabra University of Barcelona. While concerning with the possibility of enabling a joint research in both artistic and technological fields, the *Allosphere* project has also to be cited, (8). *Allosphere* is the result of 26 years of research and it allows visualizing, hearing and exploring complex multi-dimensional data. Scientifically, the *AlloSphere* can help provide insight on environments into which the body cannot venture. Artistically, the *AlloSphere* can serve as an instrument for new creations and performances fusing art, architecture, science, music, media, games, and cinema. Another related multidisciplinary application is the field at the intersection between adaptive music and games. For example, (11) presents an experimental application for individualized adaptive music for games.

Between the different technological innovations suitable for artistic purposes, *Wireless Sensor Networks* (WSNs) are used very often. For instance *Spheres and Splinters* is a new work composed by Tod Machover for hypercello, electronics, and responsive visuals that exploits audio analysis and a multitude of wireless sensors on the cello and the bow that capture how the instrument is being played. WSNs are used for personal purposes in (9) where the basic idea is to generate soundtracks for portable music players basing on the activity the user is carrying on. This way a proper music for exercise can be available to the user without the need for him to choose.

The main advantages deriving from the use of WSNs in such an application field is given by their flexibility and suitability for temporary network setups. Moreover implementation cost is cheaper than wired network even if they are more complex to configure and sometimes may be affected by surrounding events.

Advantages of using WSNs can be exploited for various purposes such as military target tracking and surveillance, natural disaster relief, biomedical health monitoring, and hazardous environment exploration and seismic sensing, (10). In military target tracking and surveillance, a WSN can assist in intrusion detection and identification. With natural disasters, sensor nodes can sense and detect the environment to forecast disasters before they occur. In biomedical applications, surgical implants of sensors can help monitor a patient's health. For seismic sensing, ad hoc deployment of sensors along the volcanic area can detect the development of earthquakes and eruptions. With the main goal to provide secure living environments for people in the world, a class of approaches refers to the exploitation of wireless sensor networks for surveillance purposes, (16).

2. RF sounding overview

This section is intended to introduce the main components of the project as summarized in figure 1. Innovations introduced and novelty will be described in detail in the next sections.

RF Sounding is an artistic installation whose aim is twofold. Indeed from one side we want to increase end users knowledge of the strength of the power emitted by their cellular phones

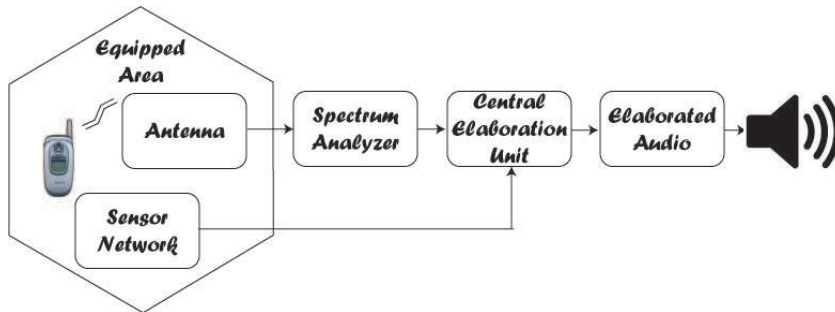


Fig. 1. RF Sounding-dataflow diagram.

with respect to the electromagnetic fields produced in the environment, on the other hand we want to provide for an artistic and interactive installation that can also be remotely joined through a web interface. It is worth noting that everything is based on the conversion of Radio Frequencies (RFs) that can be registered inside the area, into audible sounds that are spread all over the installation space through a sound diffusion system.

Here follows a brief description of the installation's architecture. Axonometric views of the project are provided in figures 2 and 3, while a block diagram is given in figure 1.

RF Sounding is built inside a hexagonal area that is accessible through at least 2 entrances equipped with gating sensors; these are intended to provide user entrance information. A loudspeaker is placed on each vertex of the hexagon in order to exploit sound spatialization. Along with the speakers there are three or more wireless sensor nodes that feed a positioning algorithm allowing to evaluate the user position and movement in the equipped area. These sensors thus allow the installation to interact with user's movements by changing lights and sounds conditions. In the center of the hexagon, at a level of 2.5-3 m from the ground, a receiving antenna is placed in order to gather all signals in the band of interest and to send them to a spectrum analyser. The analyser is linked to an elaboration unit, equipped with an audio processing board, that implements sound's elaboration and spatialization algorithms. This unit also handles the processing of the localization data obtained from the WSN. RF Sounding thus allows a user that enters the installation area bringing his switched off mobile phone, to perceive a low intensity acoustic signal that can be associated to RF signals emitted by far sources such as *Base Stations* (BSs) or other *Mobile Terminals* (MTs). On the other hand if the user inside the equipped area switches on his cellphone, he will sense a much higher acoustic signal.

3. From RF to audible frequencies

In this section we describe the process used to transform radio frequencies coming from cellular networks, into audible frequencies. The first subsection is intended to briefly review frequencies utilization and organizations in GSM and UMTS standards, the second subsection proposes an example of two cellular networks procedures that are transformed into sound events, while the third subsection goes deeper inside the translation procedure for both standards.

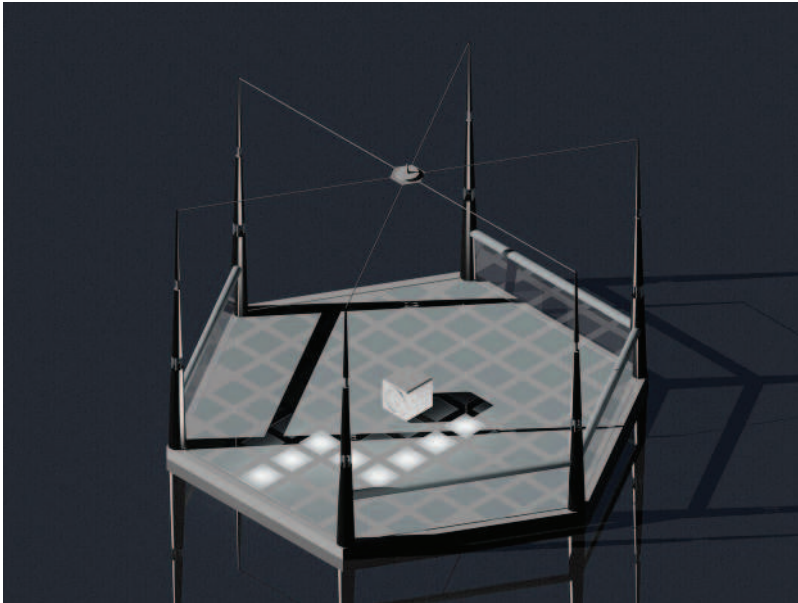


Fig. 2. RF Sounding-axonometric view.

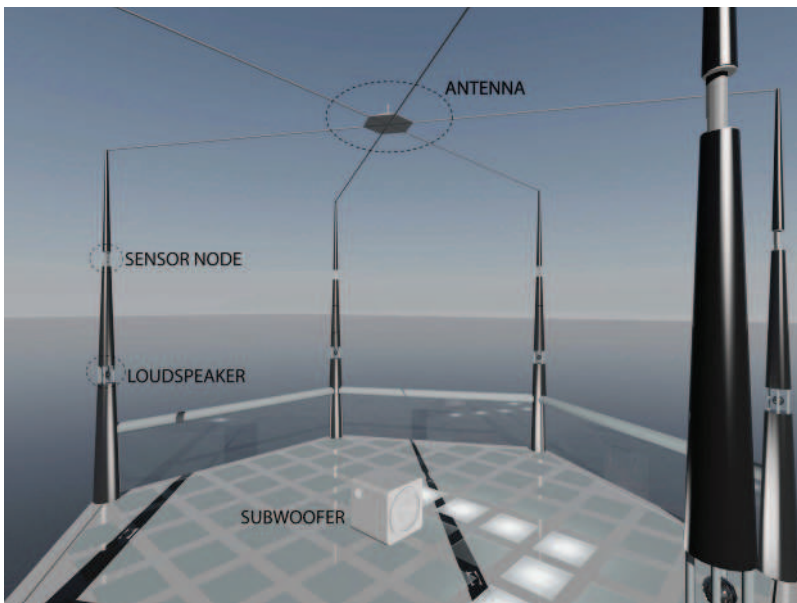


Fig. 3. RF Sounding-axonometric view with details.

System	Band	Uplink (MHz)	Downlink (MHz)	Channel number
T-GSM-380	380	380.2-389.8	390.2-399.8	dynamic
T-GSM-410	410	410.2-419.8	420.2-429.8	dynamic
GSM-450	450	450.6-457.6	460.6-467.6	259-293
GSM-480	480	479.0-486.0	489.0-496.0	306-340
GSM-710	710	698.2-716.2	728.2-746.2	dynamic
GSM-750	750	747.2-762.2	777.2-792.2	438-511
T-GSM-810	810	806.2-821.2	851.2-866.2	dynamic
GSM-850	850	824.2-849.2	869.2-894.2	128-251
P-GSM-900	900	890.0-915.0	935.0-960.0	1-124
E-GSM-900	900	880.0-915.0	925.0-960.0	975-1023, 0-124
R-GSM-900	900	876.0-915.0	921.0-960.0	955-1023, 0-124
T-GSM-900	900	870.4-876.0	915.4-921.0	dynamic
DCS-1800	1800	1710.2-1784.8	1805.2-1879.8	512-885
PCS-1900	1900	1850.2-1909.8	1930.2-1989.8	512-810

Table 1. GSM bands assignment.

It is worth noting that aesthetic considerations about these procedures are taken into account in section 6.

3.1 GSM and UMTS frequencies exploitation

The shared global use of the radio spectrum is established by ITU (International Telecommunication Union) (17), that promotes international cooperation in assigning satellite orbits, works to improve telecommunication infrastructure in the developing world and establishes worldwide standards.

We first focus on GSM, whose radio technology is specified in the 3GPPTM TS 45.-series specifications, (18), (19). We particularly refer to (19), where frequency bands of GSM systems are presented.

A GSM system may operate in 14 frequency bands as presented in table 1.

The carrier spacing, that is the channel bandwidth is of 200 KHz.

GSM-900 and GSM-1800 are the most used in Europe and we thus focus on these 2 bands of interest. It has to be noticed that GSM uses a variety of channels that are distinguished into physical and logical channels.

The method to divide up the bandwidth among as many users as possible, chosen by GSM, is a combination of Time- and Frequency-Division Multiple Access (TDMA/FDMA). FDMA divides the frequency band, which has a width of (maximum) 25 MHz, into 124 carrier frequencies. Each Base Station (BS) is assigned one or more carrier frequencies. Using a TDMA scheme each carrier frequency is divided in time, which forms logical channels. More in particular a channel number assigned to a pair of frequencies, one uplink and one downlink, is known as an Absolute Radio Frequency Channel Number (ARFCN). GSM divides up each ARFCN into 8 time slots. These 8 timeslots are further broken up into logical channels.

Logical channels can be thought of as just different types of data that is transmitted only on certain frames in a certain timeslot. Different time slots will carry different logical channels. We underline that RF Sounding does not concern with logical channels since it relies on the use of a spectrum analyser.

A first prototype of our project that has been developed on 2010, (20), used a GSM engine (Siemens TC35, (21)) that allows to check only for the downlink channel, returning information concerning the serving Base Station (BS) channel and the power received (Rx power) as well as the same parameters for adjacent BSs. With the introduction of the spectrum analyser we take into account both uplink and downlink physical channels.

GSM standard has been the first to be investigated in the project, subsequently we deepened problems and issues arising while considering 3G networks.

UMTS is an evolution of GSM as well as a complete system architecture, offering substantially higher data rates with the main goal of delivering multimedia services in the mobile domain. The process of reserving and allocating frequency spectrum for 3G networks began on 1992 during the World Administrative Radio Conference and ended on 1997 with Resolution 212 adopted at the World Radiocommunication Conference (Geneva, Switzerland), that endorsed the bands specifically for the International Mobile Telecommunications-2000 (IMT-2000) specification. According to WARC-92 "the bands 1885-2025 MHz and 2110-2200 MHz are intended for use on a worldwide basis, by administrations wishing to implement International Mobile Telecommunications-2000 (IMT-2000)". Basing on (22) and (23), UMTS frequencies can be summarized as follows:

- 1920-1980 and 2110-2170 MHz Frequency Division Duplex (FDD, W-CDMA) Paired uplink and downlink, channel spacing is 5 MHz and raster is 200 kHz. An Operator needs 3 - 4 channels (2x15 MHz or 2x20 MHz) to be able to build a high-speed, high-capacity network.
- 1900-1920 and 2010-2025 MHz Time Division Duplex (TDD, TD/CDMA) Unpaired, channel spacing is 5 MHz and raster is 200 kHz. Tx and Rx are not separated in frequency.
- 1980-2010 and 2170-2200 MHz Satellite uplink and downlink.

CDMA (Code Division Multiple Access) is a spread spectrum multiple access technique where Data for transmission is combined via bitwise XOR (exclusive OR) with pseudorandom code whose rate is much higher than the data to be transmitted. Moreover UMTS supports both frequency division and time division duplexing mode.

Carrier frequencies are designated by a UTRA Absolute Radio Frequency Channel Number (UARFCN). The general formula relating frequency to UARFN is: $UARFCN = 5 * (\text{frequency in MHz})$.

3.2 Cellular network procedures examples

In this subsection we present two of the procedures that characterize GSM cellular communications and that are transformed into an audible sound by RF Sounding.

The first procedure is related to the switch on of a MT inside the equipped area. In absence of previous information, the MT starts scanning the radio channels in the service band in order to find the carrier frequencies emitted by each cell with constant power. Once this procedure

is completed the MT has a list of all possible BS to which it can connect, and it chooses the BS characterized by the highest receive power (top of the list). Once the carrier frequency is known, the local oscillator (LO) of the mobile terminal is tuned to this frequency through a phase locked loop operating on a periodically emitted Frequency Burst (FB) (40) carried by the Frequency Correction Channel (FCCH) of the Broadcast Control Channel (BCCH) of the GSM signalling frame structure. The information travelling through the FCCH is transmitted at the maximum power since it has to be received by all terminals. This procedure takes place even in absence of mobile terminals and this aspect is exploited in the installation, after a proper elaboration, as a persistent basic element. This basilar aspect represents the starting point of the real performance.

Another interesting procedure is the reception of a call. This event is preceded by a connection sequence that sends a signalling message on a common channel (paging channel), through which the MT is warned about the incoming call. The MT recognizes its code on the paging channel and it makes a random access on the RACH (Random Access CHannel) using the maximum allowed power, since it does not know its distance from the serving BS. If this procedure is successful (i.e. there are not collisions with other transmitting MTs), the network transmits through the AGCH (Access Grant CHannel) mapped on a beacon frequency. The remaining procedure for the prearrangement of call reception is achieved through SDCCH (Stand alone Dedicated Control CHannel). Power control is another procedure that is activated over the SDCCH in order to adapt the transmission power to the distance between the MT and the BS. Other procedures are activated over the SDCCH and they are completed when a TCH (Traffic CHannel) is assigned for the effective phonic communication. Only at this time the MT rings and this is an interesting aspect to be considered in the procedure of translation to audible frequencies. Indeed we have to adapt this fast mechanism to the possibility of our auditory system that can be affected by temporal masking when the ear is stimulated by two successive sounds separated by less than 200 ms, (24), (25).

3.3 Radio frequencies to audible sounds translation procedure

In this subsection we intend to present the process of converting radio frequencies into audible sounds.

First of all we have to keep in mind that the human auditory system is able to sense a range of frequencies between 20 Hz and 20 KHz, even if the upper limit tends to decrease with age and thus most adults are unable to hear above 16 kHz. Moreover, the ear does not respond to frequencies below 20 Hz but signals with lower frequencies can be perceived through body's sense of touch, if they are characterized by a certain level of intensity.

If we just take into account GSM, it is already evident that the bandwidth offered by GSM cellular communication is much larger than the one defined by the human hearing range. Indeed GSM-900 offers a 25 MHz uplink and 25 MHz downlink, GSM-1800 offers a bandwidth of 75 MHz both in uplink and downlink, the theoretical bandwidth of the human ear is just equal to 19980 Hz, that is 0.01998 MHz. If we also take into account that GSM-900 and GSM-1800 provide respectively 124 and 374 channels, the total number of channels offered by the GSM standard is 498, each one of them corresponding to a carrier frequency. In the simplified hypothesis of just GSM-900 and GSM-1800 the best solution would be to

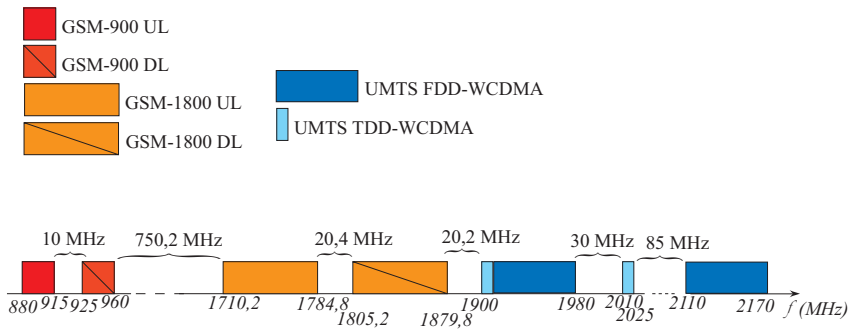


Fig. 4. GSM and UMTS frequencies allocation.

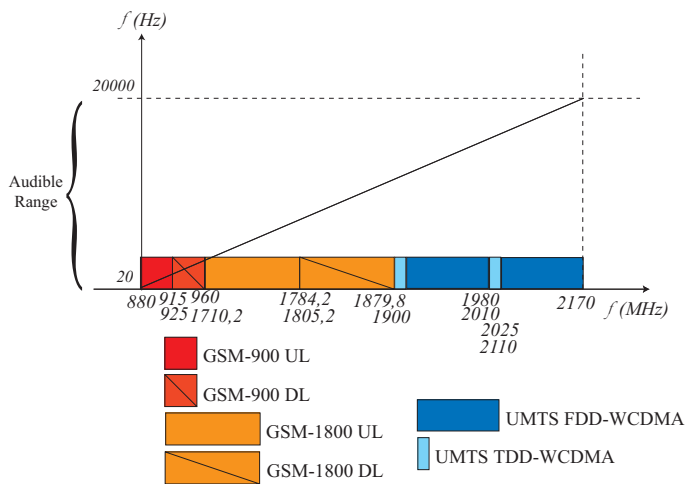


Fig. 5. Assumed relation between audible band and GSM and UMTS frequencies.

convert each carrier frequency into an audible frequency. The overall distribution of cellular frequencies for both GSM and UMTS is shown in figure 4.

By observing this figure it is evident that there are large frequency gaps that imply important gaps in an eventual linear conversion into the audible band. This can represent a problem since the audible range is very small compared to RF range, and this type of conversion would cause a noticeable loss of useful Audio Frequencies (AFs).

For this reason, in order to not lose audio bandwidth, we assumed to not consider the gaps in the RF range of interest while plotting the line corresponding to the relation between AFs and RFs, thus obtaining a conversion as roughly shown in figure 5.

As a consequence of our assumption we obtain 7 different linear relations between RFs and AFs. These relations has been computed as follows:

1. First of all we computed the equation of the line shown in figure 6 where the upper limit of the x-axis is now equal to the highest RF minus the total amount of frequency gap. In

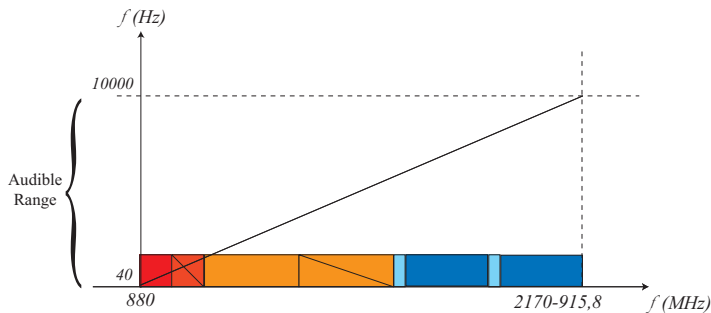


Fig. 6. Conversion procedure.

these procedure we were obliged to make our first aesthetic assumption concerning the range of AFs to be chosen. Indeed, even if the whole audible range was theoretically to be assumed for the conversion, we decided a reduction between 40 Hz and 10000 Hz, because lower frequencies are hard to be reproduced with medium quality devices, while higher frequencies than 10 KHz may result in unpleasant sensations for the human ear.

2. Once the first line equation has been found, we had to take into account that the highest AF computed for the highest RF of the first band (i.e. GSM-900 UL) is also the lowest AF corresponding to the lowest RF of the second RF band (i.e. GSM-900 DL). Indeed considering the first band we have:

$$y_{\max} = 0.02661 \cdot 10^{-3} \cdot 915 \cdot 10^{-3} - 23.38 = 0.967 \quad (1)$$

While taking into account the second band we had to force this relation:

$$y_{\max} = 0.02661 \cdot 10^{-3} \cdot 1710.2 \cdot 10^{-3} + b = 0.967 \quad (2)$$

This way we found the new constant term value.

3. The same procedure described at point 2. was followed for the other bands.

The complete list of relations found following the previously described procedure is presented below.

$$\begin{aligned} y &= 0.02661 \cdot 10^{-3}x - 23.28 && \text{GSM-900 UL} \\ y &= 0.02661 \cdot 10^{-3}x - 23.67 && \text{GSM-900 DL} \\ y &= 0.02661 \cdot 10^{-3}x - 43.61 && \text{GSM-1800 UL} \\ y &= 0.02661 \cdot 10^{-3}x - 44.15 && \text{GSM-1800 DL} \\ y &= 0.02661 \cdot 10^{-3}x - 44.69 && \text{UMTS TDD and FDD WCDMA} \\ y &= 0.02661 \cdot 10^{-3}x - 45.48 && \text{UMTS TDD-WCDMA} \\ y &= 0.02661 \cdot 10^{-3}x - 47.75 && \text{UMTS FDD-WCDMA} \end{aligned}$$

For a first implementation of RF Sounding we used Agilent HP 8592B Spectrum Analyzer shown in figure 7. The device can be connected to an elaboration unit through a IEEE 488 interface commonly called GPIB (General Purpose Interface Bus) interface, (32). Free



Fig. 7. Agilent HP 8592B Spectrum Analyzer.

Windows utilities that help making and recording research-quality measurements with GPIB-based electronic test equipment are available on the internet. A script written in C-language has been developed to ask the spectrum analyser to take and send its measures once every T seconds, where T can be chosen depending on the purpose of the installation placement and eventually the electronic music composer needs.

4. Localization through a WSN

This section is intended to provide more insight motivations and functioning of WSNs for localization purposes.

WSNs are distributed networked embedded systems where each node combines sensing, computing, communication, and storage capabilities. The nodes constituting the network are inexpensive, consisting of low power processors, a modest amount of memory, and simple wireless transceivers. These properties allowed WSNs to become very popular in recent years for applications such as monitoring, communication, and control. One of the key enabling and indispensable services in WSNs is localization (i.e., positioning), given that the availability of nodes' location may represent the fundamental support for various protocols (e.g., routing) and applications (e.g., habitat monitoring), (26).

The novelty we introduced with this project in the field of WSNs, is the application of one of their main functionalities such as localization, in the field of artistic installation.

In general there exists a variety of measurement techniques in WSN localization such as angle-of-arrival (AOA) measurements, distance related measurements and RSS profiling techniques. Distance related measurements can be further classified into one-way propagation time and round trip propagation time measurements, the lighthouse approach to distance measurements, received signal strength (RSS)-based distance measurements and time difference-of-arrival (TDOA) measurements, (27).

For a first implementation of our specific application we used sensor nodes from Memsic (originally Crossbow) called *Cricket*s, (28), (29), (30). Cricket nodes are small hardware platform consisting of a Radio Frequency (RF) transceiver, a microcontroller, and other associated hardware for generating and receiving ultrasonic signals and interfacing with a host device, figure 8. Depending on their configuration, there are two types of cricket nodes: beacons and listeners. Cricket beacons act as fixed reference points of the location system and can be attached to the ceiling or on a vertical wall depending on the application, (28), (29), while cricket listeners are attached to objects that need to obtain their location.



Fig. 8. A Cricket hardware unit.

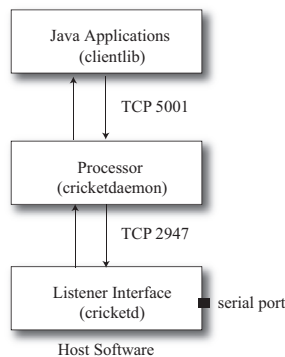


Fig. 9. Cricket software architecture.

Each beacon periodically transmits a radio frequency (RF) message containing beacon-specific information, such as beacon-ID, beacon coordinates, etc. At the beginning of the RF message, a beacon transmits a narrow ultrasonic (US) pulse that enables listeners to measure the distances to the beacons using the time difference of arrival between RF and ultrasonic signals. It is worth noting that the ultrasonic pulse does not carry any data in order to reduce beacon power consumption and ultrasonic hardware complexity.

Cricket listeners passively listen to beacon transmissions and compute distances to nearby beacons. Each listener uses these distances and the information contained in the beacon RF messages to compute their space position.

When beacons are deployed, they do not know their position. To compute beacon coordinates, it is possible to move around a cricket listener that collects distances from the beacons to itself. Using these distances, a host attached to the listener computes inter-beacon distances; the listener has to collect enough distances such that the set of computed inter-beacon distances uniquely define how the beacons are located with respect to each other.

The software for Cricket (embedded software as well as higher-layer software that runs on laptops/handhelds) is under an open source license and can be used for education, research, and commercial purposes as long as the requirements in the copyright notice are followed. The cricket embedded software is written in TinyOS (31), an open source, BSD-licensed operating system designed for low-power wireless devices. The software package includes a library to help developers create Cricket applications in Java. Cricket software architecture is shown in figure 9. At the lowest layer, `cricketd` allows a Cricket host device to access the Serial Port API to configure low-level Cricket parameters and obtain raw measurements from the Cricket hardware device. `CricketDaemon` is a server application that connects



Fig. 10. MicaZ mote.

to `cricketd` to filter and process raw Cricket measurements to infer the listener's spatial location and compute its position coordinates. The algorithm lying behind this procedure is based on the localization approach presented in (30). In a nutshell this algorithm combines the benefits of a passive mobile architecture (e.g. scalability, no need of a network infrastructure) with advantages of an active mobile system. This procedure is based on 3 main algorithms:

- *outlier rejection*: it eliminates bad distance samples;
- *extended Kalman filter* (EKF): it maintains the current and predicted device states and corrects the prediction each time a new distance sample is obtained;
- *least-squares solver* (LSQ): it minimizes the mean-squared error of a set of simultaneous non-linear equations.

Java applications may access the processed location information via the Java Cricket client library (`ClientLib`), which interfaces between the application and the `CricketDaemon`.

For our specific application we provided a software with a minimum amount of localization data elaboration based on standard deviation. This was because data coming from the underlying level appeared to be enough stable and rapid for a real time elaboration such as the one required by RF Sounding.

We also used *Memsic MicaZ* motes to provide for a wireless connection between the listener and the central elaboration unit. A MicaZ mote is shown in figure 10. Basically a MicaZ is connected to a listener, while the other one is connected to the central elaboration unit, the motes provide a transparent connection between the listener and the central elaboration unit. This way the user bringing a listener is free to move inside the installation area.

5. Open sound control protocol

In this section we want to provide for a brief overview of methods used to transmit localization data and radio spectrum information to the central elaboration unit shown in figure 1. In particular we focus on the use of *Open Sound Control Protocol* an open, transport-independent, message-based protocol developed for communication among computers, sound synthesizers, and other multimedia devices, (33), (34). OSC was originally developed, and continues to be a subject of ongoing research at UC Berkeley Center for New Music and Audio Technology (CNMAT).

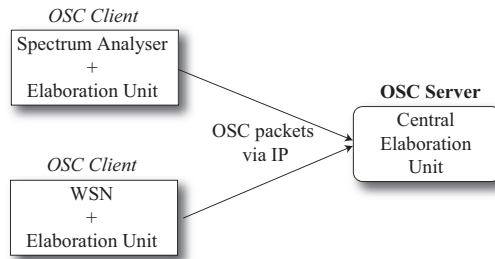


Fig. 11. Communication configuration.

OSC provides some very useful and powerful features that were not previously available in MIDI, including an intuitive addressing scheme, the ability to schedule future events, and variable data types, (35). Moreover the address space could be expanded through a hierarchical namespace similar to URL notation. Using this type of addressing allows different programs to create its own address hierarchy so that the same objects will not need the same addresses from program to program.

The main characteristic that was really appealing from our point of view is that OSC is a transport-independent protocol, meaning that it is a format for data that can be carried across a variety of networking technologies.

OSC data are basically organized into messages, which consists of the following:

- a symbolic address,
- a message name,
- the message payload.

With respect to the well known MIDI protocol, OSC is transmitted on systems with a bandwidth in the 10+ megabit/sec, that is almost 300 times faster than MIDI (31.25 kilobit/sec). Moreover precision is improved and it is much easier to work with symbolic names of objects rather than complicated mapping of channel numbers, program change numbers and controller numbers as in MIDI. On the other hand it has to be noticed that OSC can not replace MIDI, due to missing automatic connect-and play (or plug-and-play) concept, that is connected devices (via Ethernet, WLAN, Bluetooth etc) cannot scan each other and learn about each others capabilities, moreover a file format such as standard MIDI file for exchange of data does not exist.

Any application that sends OSC Packets is an OSC Client; any application that receives OSC Packets is an OSC Server. An OSC server must have access to a representation of the correct current absolute time. OSC does not provide any mechanism for clock synchronization but assumes that the two interacting systems will provide a mechanism for synchronisation. Time tags eliminate jitter introduced during transport by resynchronizing messages in a bundle and setting values for when they should take place.

Given the main advantages coming from the use of the OSC protocol we decided to assume the flexible configuration shown in figure 11.

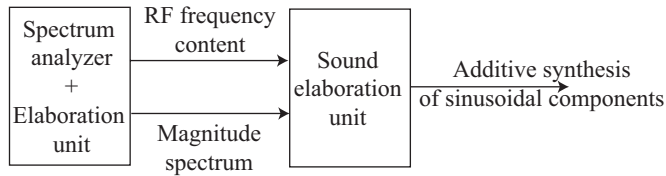


Fig. 12. Sound production for educational purposes.

The Central Elaboration Unit is equipped with Max 5, (36), a visual programming language for music and multimedia that is used in our application in order to implement sound synthesis and elaboration as well as sound spatialization as a function of RFs translation and localization data. The basic language of Max is that of a data-flow system: Max programs (called *patches*) are made by arranging and connecting building-blocks of *objects* within a *patcher*, or visual canvas. These objects act as self-contained programs (in reality, they are dynamically-linked libraries), each of which may receive input (through one or more visual "inlets"), generate output (through visual "outlets"), or both. Objects pass messages from their outlets to the inlets of connected objects.

For our purpose of communication through OSC we exploited the use of two main Max objects: `udpsend` and `udpreceive` that allow to realize the configuration shown in figure 11.

6. Sound synthesis and real time processing

In this section we provide for an overview of the main algorithms and techniques used to generate and to spread the sound all over the installation area.

A first distinction has to be done considering the possible applications of the project. Indeed it is clear that if the installation is used with educational purposes, than the sound to be spread through loudspeakers has to represent RFs behaviour as close as possible. On the other hand if the project is considered as an artistic installation rich in technological innovations, than the sound reproduced must be generated taking into account aesthetic considerations.

Let us consider the first applicative field. While using RF Sounding with educational purposes we have to reproduce RFs behaviour such that their effects should be clearly understood by the users. Having this in mind, we decided to assign each translated frequency to a sine wave, whose amplitude is thus defined by the magnitude corresponding to that frequency and computed by the spectrum analyser. Two schemes to better explain this process are presented in figures 12 and 13. The latter is referred to a particular of the procedure of conversion in the Max patch that has been developed. It has also to be noticed that the proportionality between amplitude of the audible signal and emitted power is of fundamental importance to understand how operations on a cellular phone influence the electromagnetic pollution. The effect of this procedure can be easily imagined in the case of mobile switching on and call reception procedures described in section 3.

When the installation is used for artistic purposes it becomes an interesting opportunity for modern music composers to introduce a proper signal processing on revealed RF signals. This way the installation becomes a new instrument for which new music can be composed.

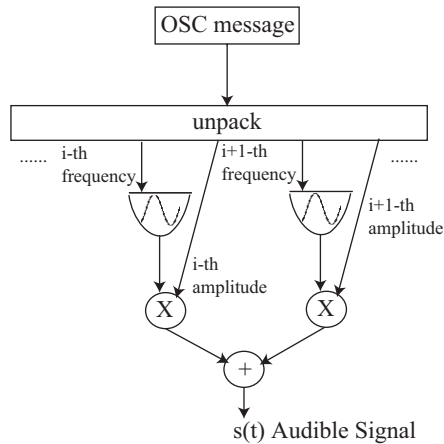


Fig. 13. Max like scheme in sound production for educational purposes.

Basing on our aesthetic point of view we decided to synthesize the sound through 3 main techniques:

1. white noise filtering with resonant filters whose center frequency is defined by revealed RFs and whose Q-factor, (37) is a function of the measured amplitudes;
2. FM parallel modulations with one carrier and one modulating signals that are added to produce the final result, (38);
3. Additive synthesis of sinusoidal components found as in the educational case plus granular synthesis, (39) where envelope and silence interval of duration are defined a a function of revealed frequencies.

It is worth noting that given a set of frequencies and values corresponding to amplitudes, these sets can be used for sound synthesis in an almost infinite number of ways including Amplitude Modulation, Phase Modulation, filtering of complex signals and so on.

We decided to introduce a random variation between the 3 methods of synthesis previously listed, maintaining the randomness between 3 and 5 minutes. It has to be noticed that the reproduced sound is pretty much variable because of different operations that are exploited by the MT or the BTSs acting on the installation area.

We also exploited the localization data coming from the WSN to produce multichannel spatialiation of sound. While taking into account this procedure, many parameters can be varied, (41), (42), (43):

- power of all the signals emitted in the environment,
- sound motion,
- number of loudspeakers involved in sound emission in a certain instant,
- crossfade envelope between two or more channels,
- residual power of loudspeakers not directly involved in main sound emission.

For what concerns spatialization algorithms it has to be noticed that our installation allows a countless number of possibilities.

A first characterization has to be done as a function of the user's motion inside the equipped area while the installation is used for educational purposes. In this case we made an analysis on the background RF Spectrum with respect to RFs produced by active operations of the user inside the area. Indeed when the spectrum is only defined by periodic downlink operations, the sound produced is not spatialized but instead it is uniformly diffused around the area with a low intensity. On the other hand when user's MT makes its own operation, the sound is spatialized such that to give the user's the impression of being followed.

Obviously when the installation is used with artistic purposes the possibilities allowed by the sound diffusion system can be exploited with a major freedom. In this case we fixed a threshold for the user's speed. Below this threshold we established the sound diffusion system to emit only a low circular spatialized sound. This is achieved by implementing the sound crossfade only between adjacent loudspeaker. The envelope is characterized by sinusoidal panning curves in order to avoid too rapid sound variations between channels. It has to be noticed that this choice requires the introduction of at least one element of randomness on sound speed in order to avoid an unpleasant "dance" effect on sound motion.

When the user is moving faster we assumed to spatialize the sound with the following methods:

- diagonal motion between single loudspeakers,
- motion between pairs of loudspeakers,
- slow motion between three loudspeakers with respect to a single loudspeaker or a pair of loudspeakers such that to concentrate sound intensity in a certain point in the space,
- offset variation between loudspeakers not directly involved in sound emission.

It is worth noting that all previously described mechanisms are automated as a function of user position and motion, but also a random element is introduced in order to vary the performance result.

We want also to underline that although the installation is flexible and can be realized both indoor and outdoor, it requires the study of the impulse response of the environment and sound diffusion system characteristics.

7. Web interface

The project provides a Web interface in order to provide for a widespread diffusion of the installation's acoustic results as well as to allow the users enjoying the equipped space to record their performance and eventually use the result for instance as a ring tone. This is possible by leaving the MT just below the antenna connected to the spectrum analyser for a few seconds. This way the installation produces a sort of audio signature that is quite specific for each MT since both transmission power and RFs change as a function of the mobile operator furnishing the service as well as the specific brand and model of the MT itself.

This generated ring tone is associated to the user who can download it by logging in the installation website. This operation can be done in the installation area that is provided with a WiFi access.

8. Conclusions and future works

In this chapter an innovative project integrating technologies and experimental music has been described. RF Sounding is based on the most recent technological innovations that are exploited in an unusual fashion and it focuses on the creation of an interactive installation with a double aim, to increase users' awareness of the spectral occupancy in the cellular networks bands and to provide for a spectral phenomena aesthetic elaboration in order to produce a sounding experience.

A lot of improvements can be achieved on this project. From the WSN point of view it is important to develop a passive localization system through the exploitation of RSSI information as in multistatic RADAR, (44), (45), this way the user is relieved of bringing an external node inside the equipped area. Still concerning the technological aspect we want to make reproducible also other communication standards such as Wifi, WiMax, Zigbee and Bluetooth etc.

On the artistic point of view, we want to improve the installation's adaptability and versatility to different musical contexts, simplicity and rapidity in bringing changes in events, stability of the entire system and overall sound quality.

9. References

- [1] Anna Trifonova and Letizia Jaccheri and Kristin Bergaust, "Software Engineering Issues in Interactive Installation Art" *International Journal on Arts and Technology (IJART)*, volume 1, number 1, pages 43-65, 2008.
- [2] Trifonova, Anna and Brandtsegg, Øyvind and Jaccheri, Letizia, "Software engineering for and with artists: a case study", in *Proceedings of the 3rd international conference on Digital Interactive Media in Entertainment and Arts*, pages 190-197, 2008.
- [3] Salah Uddin Ahmed and Letizia Jaccheri and Samir M'kadmi, "Sonic Onyx: Case Study of an Interactive Artwork." *ArtsIT 2009: International Conference on Arts & Technology, Springer Lecture Notes of ICST*, 2009
- [4] Li, Wu-Hsi, "Musicpainter : a collaborative composing environment", Master's Thesis, Massachusetts Institute of Technology. Dept. of Architecture. Program in Media Arts and Sciences, 2008.
- [5] Mihir Sarkar, *TablaNet: a Real-Time Online Musical Collaboration System for Indian Percussion*, Master's Thesis, MIT Media Lab, Aug. 2007.
- [6] MIT Media Lab, *Opera of the Future*, Available: <http://www.media.mit.edu/research/groups/opera-future>.
- [7] Jordà, Sergi, "The reactable: tangible and tabletop music performance", in *Proceedings of the 28th of the international conference extended abstracts on Human factors in computing systems*, pages 2989-2994, Atlanta, Georgia, USA, 2010.
- [8] Höllerer, Tobias and Kuchera-Morin, JoAnn and Amatriain, Xavier, "The allosphere: a large-scale immersive surround-view instrument" in *Proceedings of the 2007 workshop on*

- Emerging displays technologies: images and beyond: the future of displays and interaction*, San Diego, California, 2007.
- [9] Robert Jacobs, Mark Feldmeier, Joseph A. Paradiso, "A Mobile Music Environment Using a PD Compiler and Wireless Sensors" *the Proc. of the 8th International Conference on New Interfaces for Musical Expression*, 2008.
- [10] I. F. Akyildiz and W. Su and Y. Sankarasubramaniam and E. Cayirci, "Wireless sensor networks: a survey" in *Computer Networks*, Amsterdam, Netherlands: 1999, (38) 4: 393–422, Year 2002.
- [11] Eladhari, Mirjam and Nieuwdorp, Rik and Fridenfalk, Mikael, "The soundtrack of your mind: mind music - adaptive audio for game characters", in *Proceedings of the 2006 ACM SIGCHI international conference on Advances in computer entertainment technology*, 2006.
- [12] Xavier Amatriain and JoAnn Kuchera-Morin and Tobias Hollerer and Stephen Travis Pope, "The AlloSphere: Immersive Multimedia for Scientific Discovery and Artistic Exploration" *Journal IEEE Multimedia*, IEEE Computer Society, vol. 16, pages 64–75, Los Alamitos, CA, USA, 2009.
- [13] A. K. Turza, "Dense, Low-Power Environmental Monitoring for Smart Energy Profiling" *Bachelor Thesis*, 2009.
- [14] Paradiso, Joseph and Gips, Jonathan and Laibowitz, Mathew and Sadi, Sajid and Merrill, David and Aylward, Ryan and Maes, Pattie and Pentland, Alex, "Identifying and facilitating social interaction with a wearable wireless sensor network" *Journal of Personal and Ubiquitous Computing*, Springer London, volume 14, pages 137–152, 2010.
- [15] Aylward, Ryan and Paradiso, Joseph A., "A compact, high-speed, wearable sensor network for biomotion capture and interactive media" in *Proceedings of the 6th international conference on Information processing in sensor networks*, pages 380–389, Cambridge, Massachusetts, USA, 2007.
- [16] R. Alesii, F. Graziosi, G. Gargano, L. Pomante, C. Rinaldi, "WSN-Based Audio Surveillance Systems" in *Proceedings of the European Computing Conference*, Lecture Notes in Electrical Engineering Ed. Springer US, 2009.
- [17] ITU official site, <http://www.itu.int/en/Pages/default.aspx>.
- [18] ETSI, 3GPP TR 45.050 version 10.0.0 Release 10 "Digital cellular telecommunications system (Phase 2+); Background for RF Requirements (3GPP TR 45.050 version 10.0.0 Release 10)", *Technical report*, April 2011.
- [19] ETSI TS 145 005 V10.1.0, "Digital cellular telecommunications system (Phase 2+); Radio transmission and reception (3GPP TS 45.005 version 10.1.0 Release 10)", *Technical report*, June, 2011 .
- [20] C. Rinaldi, L. Pomante, R. Alesii, F. Graziosi, "RF sounding" in *Proceedings of the 8th ACM Conference on Embedded Networked Sensor Systems*, pages 363-364, Zürich, Switzerland, 2010.
- [21] Siemens TC35, Available at: <http://www.alldatasheet.com/view.jsp?sSearchword=TC35>.
- [22] ETSI TS 125 104 V10.1.0, "Universal Mobile Telecommunications System (UMTS); Base Station (BS) radio transmission and reception (FDD) (3GPP TS 25.104 version 10.1.0 Release 10)" *Technical Specification*, May, 2011.
- [23] ETSI TS 125 105 V10.3.0, "Universal Mobile Telecommunications System (UMTS); Base Station (BS) radio transmission and reception (TDD) (3GPP TS 25.105 version 10.3.0 Release 10)", *Technical Specification*, July 2011.

- [24] F. Alton Everest, Ken Pohlmann, *Master Handbook of Acoustics*, Fifth Edition, Ed. Mac Graw-Hill, 2009.
- [25] Albert S. Bregman, *Auditory Scene Analysis: The Perceptual Organization of Sound*, Cambridge, MA: MIT Press, 1990.
- [26] Stefano Tennina, Marco Di Renzo, Fabio Graziosi, Fortunato Santucci, "ESD: A Novel Optimization Algorithm for Positioning Estimation of WSNs in GPS-denied Environments – From Simulation to Experimentation" in *International Journal of Sensor Networks*, Vol. 6, Pages 131-156, Inderscience Publishers, 2009 .
- [27] Guoqiang Mao, Baris Fidan and Brian D.O. Anderson, "Wireless sensor network localization techniques" in *Journal of Computing Networks*, Vol. 51, Pages 2529-2553, Elsevier North-Holland, Inc., 2007.
- [28] Nissanka Bodhi Priyantha, "The Cricket Indoor Location System" *PhD Thesis*, Massachusetts Institute of Technology, June 2005.
- [29] Nissanka B. Priyantha, Anit Chakraborty and Hari Balakrishnan, "The Cricket Location-Support System" in *Proceeding of 6th ACM MOBICOM*, Boston, MA, August 2000.
- [30] Adam Smith, Hari Balakrishnan, Michel Goraczko, and Nissanka Priyantha, "Tracking Moving Devices with the Cricket Location System" in *Proceedings of 2nd USENIX/ACM MOBISYS Conference*, Boston, MA, June 2004 .
- [31] TinyOS home page, Available at: <http://webs.cs.berkeley.edu/tos/>.
- [32] International Standard, "IEC/IEEE Standard for Higher Performance Protocol for the Standard Digital Interface for Programmable Instrumentation - Part 1: General (Adoption of IEEE Std 488.1-2003)" in *IEC 60488-1 First edition 2004-07; IEEE 488.1*, pp.1-158, 2004.
- [33] Wright, M. and A. Freed, "Open SoundControl: A New Protocol for Communicating with Sound Synthesizers" in *International Computer Music Conference Thessaloniki, Hellas*, 1997.
- [34] Open Sound Control Official Website, Available at: <http://opensoundcontrol.org/> .
- [35] Angelo Fraietta, "Open Sound Control: Constraints and Limitations" in *Proceedings of the 6th International Conference of New Interfaces for Musical Expression*, 5-7 June 2008, Italy .
- [36] Max 5 Help and Documentation, available at: <http://cycling74.com/docs/max5/vignettes/intro/docintro.html>
- [37] Sanjit K. Mitra, *Digital Signal Processing: A Computer Based Approach*, 4th Edition, McGraw Hill International Edition, 2011.
- [38] J. Chowning, "The Synthesis of Complex Audio Spectra by Means of Frequency Modulation" in *Journal of the Audio Engineering Society*, vol 21, 1973 .
- [39] Curtis Roads, *The Computer Music Tutorial*, Cambridge: The MIT Press, 1996.
- [40] U. S. Jha, "Acquisition of frequency synchronization for GSM and its evolution systems" in *Personal Wireless Communications*, 2000 IEEE International Conference on Volume , Issue , 2000 Page(s):558 - 562.
- [41] Vidolin A. "Spazi fisici e spazi virtuali nella musica elettroacustica", in *I Quaderni della Civica Scuola di Musica*, special number dedicated to Music Space and Architecture, y. 13, n. 25, pp.58-63, 1995. Revised and expanded in *Ejecutar el espacio.*, Azzurra, A. VII, n.13-15 Istituto Italiano di Cultura a Cordoba, 2000.

- [42] Stockhausen K., "Musik im Raum", 1958, italian translation "Musica nello spazi, in *La Rassegna Musicale*, 32(4), 1961.
- [43] Rizzardi V., "L'impiego dello spazio" in *Il Suono e lo Spazio*, Catalogo, RAI Sede regionale per il Piemonte, Turin, June 1987.
- [44] Pavel Bezouseki, Vladimír Schejbal, "Bistatic and Multistatic Radar Systems" in *Radioengineering*, vol 17, n. 3, September, 2008.
- [45] Victor S. Chernyak, "Fundamentals of Multisite Radar Systems: Multistatic Radars and Multistatic Radar Systems" *CRC Press*, September 1998.



Management of Technological Innovation in Developing and Developed Countries

Edited by Dr. HongYi Sun

ISBN 978-953-51-0365-3

Hard cover, 312 pages

Publisher InTech

Published online 21, March, 2012

Published in print edition March, 2012

It is widely accepted that technology is one of the forces driving economic growth. Although more and more new technologies have emerged, various evidence shows that their performances were not as high as expected. In both academia and practice, there are still many questions about what technologies to adopt and how to manage these technologies. The 15 articles in this book aim to look into these questions. There are quite many features in this book. Firstly, the articles are from both developed countries and developing countries in Asia, Africa and South and Middle America. Secondly, the articles cover a wide range of industries including telecommunication, sanitation, healthcare, entertainment, education, manufacturing, and financial. Thirdly, the analytical approaches are multi-disciplinary, ranging from mathematical, economic, analytical, empirical and strategic. Finally, the articles study both public and private organizations, including the service industry, manufacturing industry, and governmental organizations. Given its wide coverage and multi-disciplines, the book may be useful for both academic research and practical management.

How to reference

In order to correctly reference this scholarly work, feel free to copy and paste the following:

Claudia Rinaldi, Fabio Graziosi, Luigi Pomante and Francesco Tarquini (2012). RF Sounding: Generating Sounds from Radio Frequencies, Management of Technological Innovation in Developing and Developed Countries, Dr. HongYi Sun (Ed.), ISBN: 978-953-51-0365-3, InTech, Available from:
<http://www.intechopen.com/books/management-of-technological-innovation-in-developing-and-developed-countries/rf-sounding-generating-sounds-from-radio-frequencies>

INTECH
open science | open minds

InTech Europe

University Campus STeP Ri
Slavka Krautzeka 83/A
51000 Rijeka, Croatia
Phone: +385 (51) 770 447
Fax: +385 (51) 686 166
www.intechopen.com

InTech China

Unit 405, Office Block, Hotel Equatorial Shanghai
No.65, Yan An Road (West), Shanghai, 200040, China
中国上海市延安西路65号上海国际贵都大饭店办公楼405单元
Phone: +86-21-62489820
Fax: +86-21-62489821

© 2012 The Author(s). Licensee IntechOpen. This is an open access article distributed under the terms of the [Creative Commons Attribution 3.0 License](#), which permits unrestricted use, distribution, and reproduction in any medium, provided the original work is properly cited.