Combining GSM and WVoIP Technologies

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Abstract

This paper describes an enhancement to the existing cellular phone system which will provide improved communication capabilities on any campuses. It will allow individuals to receive calls made to their office phone on their personal cell phones, while still permitting the normal use of the cell phone. This innovative system provides the ability for users to keep in touch no matter where they are on campus.

This new phone system will utilize Wireless Voice over Internet Protocol as an addition to a regular GSM technology, without incurring any additional costs from their cellular provider. This can be achieved by reconfiguring the current campus Internet wireless facility and designing a new cellular phone. The review of related literature for this project has already been done.

Introduction

Over the years, the world has become a smaller place to live in due to the different kinds of improvement in the world of technology. One of the greatest inventions that science has offered us is the telephone which has been the best means of communication in the past couple of centuries. However, the telephone itself has been upgraded to offer the societies better services and means of contact.

Many factors have played a major role in helping the telephone technology to achieve progress and success, but it wasn't until wireless technology was introduced that the telephone business hit the market. Mobile phones, telephones that utilize wireless technology to connect to the networks and which are also called cellular phones, have been really popular nowadays. You can always get in touch with a person wherever, whenever and whatever he is doing. These phones have been the main reason for our world to progress rapidly.

As cellular phones started to get more and more popular, the Internet world started to become more public and available to everyone. Similar to any technological invention, the cost of a product decreases with time since it would be offered in the public market and more people tend to buy it. Accordingly, the cost of Internet started to decrease and more and more companies started to realize the benefits that they could get from it. This process led to the notion of embedding Internet within a phone and coming up with a Voice over IP phone. In the beginning, these phones gained their popularity on the corporate scale, but later on, different institutions started to demand this product as a reasonable means of communication to organize their work without any extra-ordinary expenses. [1]

Despite the expansion and the increase in the use of VoIP technology with time, there was still demand for equipment and products that would offer cheaper and more mobile means of calling. This demand resulted in the discovery of Wireless of VoIP (known as WVoIP) technology which offered its users both the advantages and benefits of VoIP technology in addition to the amazing mobility, productivity and ease that wireless technology supplies them with. [2]

If WVoIP has so many benefits, then why isn't it the dominating technology in our current time? The answer lies in the fact that there is one major obstacle that faces it and that is its restrictedness within a closed network. Since WVoIP depends on the Internet protocol to function, it is useless when it has to be used with the connection offered by the ordinary GSM service providers. It is at this present state when an idea has come up to merge WVoIP and GSM technology to come up with a mobile phone that would combine both technologies.

Problem Statement

Currently, the rigorous schedule of classes and administrative tasks in the University of Hartford keep faculty and staff constantly on the move. This creates a gap in communication which can't be filled by landline phones currently in use. In addition, these devices require expensive maintenance. This new technology will eliminate the need for the purchase or the maintenance of such devices.

The solution lies in the fact that existing cellular phones (the regular cellular phones, currently being used by the staff) have to be enabled to perform a dual task. This means that the same device can receive a call which has been dialed to his official phone.

Building a new network to provide to this feature seems absurd monetarily. This project seeks to make use of the WAN (Wide Area Network) currently in place to cater for connectivity requirements and the concept of IP (Internet Phone). This can be achieved by installing a secondary transceiver capable of receiving the WAN signal. This antenna would be connected to an electronic circuit which would serve as an interface between the receiving antenna and the rest of the communication circuitry, which performs the remaining tasks of conventional two way communication. The phone call being made will be converted to an Internet call by an ATA (Analog Telephone Adaptor). The official phone number will be a reference number for a unique IP address assigned to the cellular phone.

Thus, this will provide the faculty and staff total mobility around the campus eliminating the scenario of missing an urgent call. It will lead to a much more dynamic and efficient work force.

Communications Principles

Each mobile uses a separate, temporary radio channel to talk to the cell site. The cell site talks to many mobiles at once, using one channel per mobile. Channels use a pair of frequencies for communication—one frequency (the forward link) for transmitting from the cell site and one frequency (the reverse link) for the cell site to receive calls from the users. Radio energy dissipates over distance, so mobiles must stay near the base station to maintain communications. The basic structure of mobile networks includes telephone systems and radio services. Where mobile radio service operates in a closed network and has no access to the telephone system, mobile telephone service allows interconnection to the telephone network. [3]

Cellular Concept

Interference problems caused by mobile units using the same channel in adjacent areas proved that all channels could not be reused in every cell. Areas had to be skipped before the same channel could be reused. Even though this affected the efficiency of the original concept, frequency reuse was still a viable solution to the problems of mobile telephony systems.

Engineers discovered that the interference effects were not due to the distance between areas, but to the ratio of the distance between areas to the transmitter power (radius) of the areas. By reducing the radius of an area by 50 percent, service providers could increase the number of potential customers in an area fourfold. Systems based on areas with a one-kilometer radius would have one hundred times more channels than systems with areas 10 kilometers in radius. Speculation led to the conclusion that by reducing the radius of areas to a few hundred meters, millions of calls could be served.

The cellular concept employs variable low-power levels, which allow cells to be sized according to the subscriber density and demand of a given area. As the population grows, cells can be added to accommodate that growth. Frequencies used in one cell cluster can be reused in other cells. Conversations can be handed off from cell to cell to maintain constant phone service as the user moves between cells. [3]

Cells

A cell is the basic geographic unit of a cellular system. The term cellular comes from the honeycomb shape of the areas into which a coverage region is divided. Cells are base stations transmitting over small geographic areas that are represented as hexagons. Each cell size varies depending on the landscape. Because of constraints imposed by natural terrain and man-made structures, the true shape of cells is not a perfect hexagon. [3]

Clusters

A cluster is a group of cells. No channels are reused within a cluster. Figure 1 illustrates a seven-cell cluster.

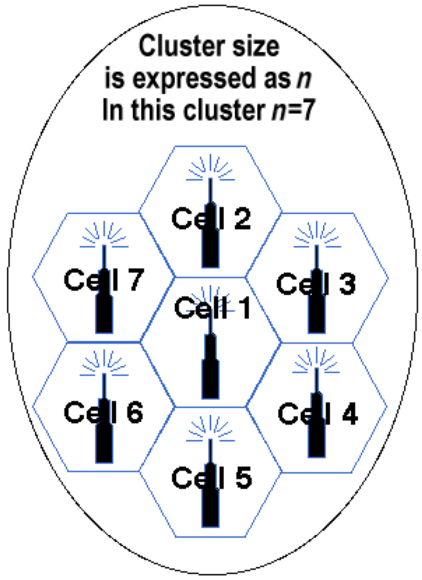

Figure 1 - Illustration of the Cell Concept

Frequency Reuse

Because only a small number of radio channel frequencies were available for mobile systems, engineers had to find a way to reuse radio channels to carry more than one conversation at a time. The solution the industry adopted was called frequency planning or frequency reuse. Frequency reuse was implemented by restructuring the mobile telephone system architecture into the cellular concept.

The concept of frequency reuse is based on assigning a group of radio channels used within a small geographic area to each cell. Cells are assigned a group of channels that is completely different from neighboring cells. The coverage area of cells is called the footprint. This footprint is limited by a boundary so that the same group of channels can be used in different cells that are far enough away from each other so that their frequencies do not interfere. An illustration is shown in figure 2.

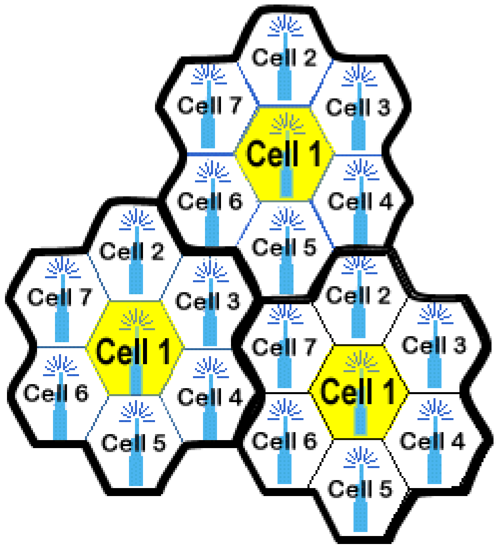

Figure 2 - foot prints of cells

Cells with the same number have the same set of frequencies. Here, because the number of available frequencies is 7, the frequency reuse factor is 1/7. That is, each cell is using 1/7 of available cellular channels. [3]

Saving bandwidth in the signal coding process

Segmentation: Given that the speech organs are relatively slow in adapting to changes, the filter parameters representing the speech organs are approximately constant during 20ms. For this reason, when coding speech in, a block of 20 ms is coded into one set of bits. In effect, it is similar to sampling speech at a rate of 50 times per second instead of the 8,000 used by A/D conversion.

Cell Splitting

As a service area becomes full of users, this approach is used to split a single area into smaller ones. In this way, urban centers can be split into as many areas as necessary to provide acceptable service levels in heavy-traffic regions, while larger, less expensive cells can be used to cover remote rural regions. Figure 3 describes this technique.

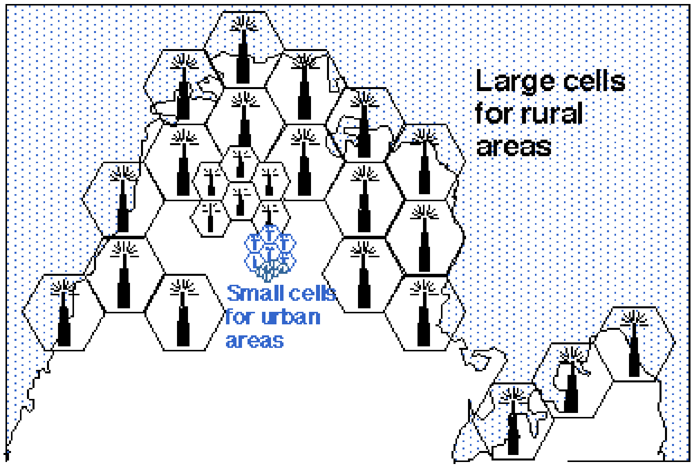

Figure 3 - Cell Splitting

Handoff

As adjacent areas do not use the same radio channels, a call must either be dropped or transferred from one radio channel to another when a user crosses the line between adjacent cells. Because dropping the call is unacceptable, the process of handoff was created. Handoff occurs when the mobile telephone network automatically transfers a call from radio channel to radio channel as a mobile handset crosses adjacent cells. Figure 4 shows this process:

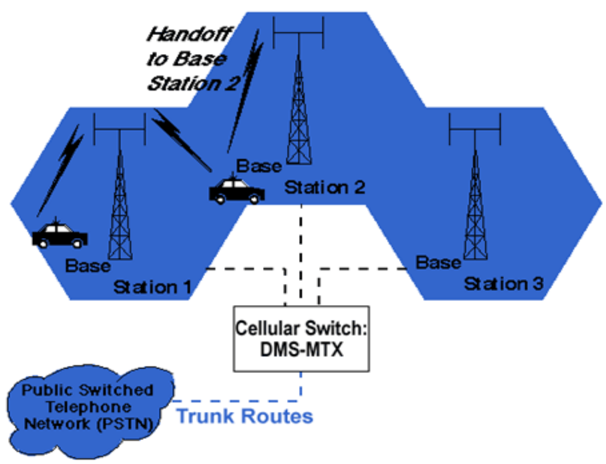

Figure 4 – Handoff to Base Stations

During a call, two parties are on one voice channel. When the mobile unit moves out of the coverage area of a given cell site, the reception becomes weak. At this point, the cell site in use requests a handoff. The system switches the call to a stronger-frequency channel in a new site without interrupting the call or alerting the user. The call continues as long as the user is talking, and the user does not notice the handoff at all.

Narrowband Analog Mobile Phone Service (NAMPS)

The NAMPS concept uses frequency division to get 3 channels in the AMPS 30-kHz single channel bandwidth. NAMPS provides 3 users in an AMPS channel by dividing the 30-kHz AMPS bandwidth into 3 10-kHz channels. This increases the possibility of interference because channel bandwidth is reduced.

The Cell Site

The term "cell site" is used to refer to the physical location of radio equipment that provides coverage within a cell. A list of hardware located at a cell site includes power sources, interface equipment, radio frequency transmitters and receivers, and antenna systems.

Mobile Subscriber Units (MSUs)

The mobile subscriber unit consists of a control unit and a transceiver that transmits and receives radio transmissions to and from a cell site. The following three types of MSUs are available:

- the mobile telephone (typical transmit power is 4.0 watts)
- the portable (typical transmit power is 0.6 watts)
- the transportable (typical transmit power is 1.6 watts)
- The mobile telephone is installed in the trunk of a car, and the handset is installed in a convenient location to the driver. Portable and transportable telephones are hand-held and can be used anywhere. The use of portable and transportable telephones is limited to the charge life of the internal battery [3].

VoIP Background

The cost of communication is a major issue in a world that relies heavily on this commodity. We are in a constant race to find cheaper and more efficient means of communication. Rapid developments over the past decade in the area of digital networking and communication culminated in Voice over Internet Protocol (VOIP). VoIP was first introduced in 1991 and has expanded rapidly to unite the telephony and data worlds. Its revolutionary approach as compared to ordinary telephone operation has made it possible for the world to have cheaper communication on a wide variety of networks with many new and exciting features such as IP multicast conferencing, and "voice web browsing". Figure 5 portrays the network over which VOIP technology works on. This greatly enhanced the popularity and success of VoIP which has become indispensable in the corporate world where corporations have the necessary bandwidths to justify it [4].

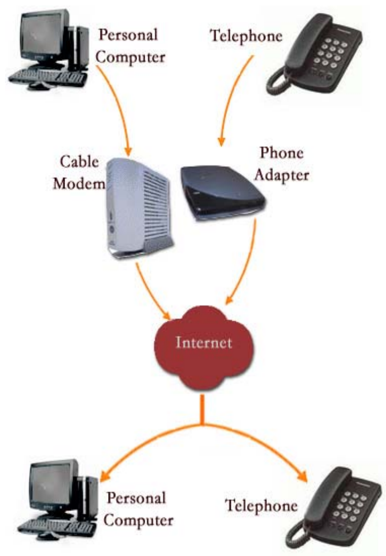

Figure 5 – Connectivity over the Internet

Moreover, IP phones have been around for a fair period of time. They work on the principle of reserving certain addresses or locations on the Internet by referring to unique numbers known as IP addresses. IP phones can be used similarly as normal telephones except that they use an RJ-45 connector and have to be connected to an Internet router. They include all the necessary hardware and software to place a call. The market is also currently providing us with WiFi enabled IP phones which rely on WiFi hot spots to send and receive the voice signals [5]. Figure 6 [6] represents a picture of a VoIP phone and figure 7 [7] represents a WVoIP phone.

 Figure 6 – VoIP Phone	 Figure 7 – Wireless VoIP Phone

The following briefly explains how the process of a VoIP phone call works:

1. The VoIP phone gets picked up and a signal is sent to the Analog Telephone Adapter (ATA) which is an Analog to Digital and Digital to Analog converter.

2. The ATA sends back a dial tone which allows you to dial the number.
3. The ATA converts the tones into digital data which is sent to the call processor where its format is checked.
4. In the call processor, the phone number is translated into an IP address.
5. The soft switch in the call processor connects the two ATAs together.
6. A session is established between both parties and packets are transmitted back and forth when data is to be sent.

The process is also represented in Figure 8 which portrays the different steps that occur within a VoIP phone call.

Moreover, the VoIP process requires tools for coding and decoding the signal from analog to digital and vice versa. This is done by using codecs. These tools work according to the transmission rule: "Send Data Only When Someone Needs to Talk". Some types of codecs could be the G.711, G.723 or G.729. How can we connect these codecs to each other? This is performed by using protocols that define the ways in which devices such as codecs connect to each other and to the network using VoIP. Recently, the H.323 family of protocols is beginning to replace the H.320 videoconference standard due to their advantage in delivering audio and video traffic. The main problem and challenge remains in being able to come up with one standard universal protocol for VoIP.

The world is leaning towards the usage of voice and data switches that would allow the transmission of voice to be in the form of packets. These switches can be of many types varying from transition switches to office switches or even remote switches that work for the customer's benefit. Thus, the key to achieve optimum network performance is the ability to handle time-sensitive voice traffic. VoIP is expanding little by little and more companies have started to offer this new technology that the customers are getting to experience as a method of reducing the telephone costs and merge their networks over a single protocol set.

Wireless VoIP Background

It is the flexibility, cheapness and the exciting new features that have rendered VoIP successful. Given the popularity of wireless technology, the next logical step is the development of wireless VoIP. WVoIP can offer the same features as VoIP and in addition would allow us to transfer data between our phone and many digital devices that can be hooked up to the Internet. This phenomenon is shown in Figure 9.

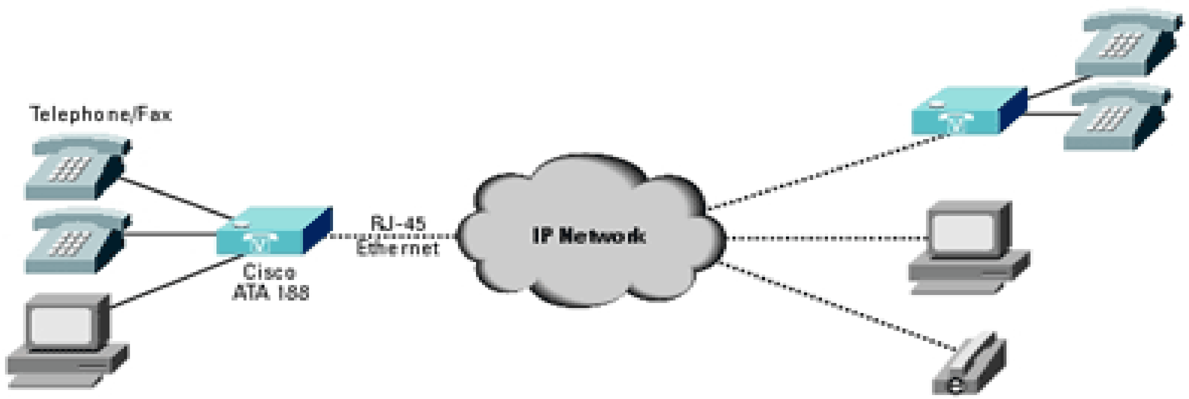

Figure 8 – Data Transfer in an IP Network

Wireless VoIP phones are much identical to traditional cordless phones; however, they depend on WiFi 802.11 wireless connectivity and other hardware to allow the Internet callers to enjoy the wireless technology while they are hooked up with the Internet service providers. 802.11 is an evolving family of specifications for wireless local area networks (WLANs) developed by a working group of the Institute of Electrical and Electronics Engineers (IEEE). There are several specifications in the family and new ones are occasionally added [8].

The current market is supporting wireless VoIP applications and a huge boost is expected in this field. Predictions by the U.S. Research Firm In-stat say that 73 percent of worldwide VoIP lines and 66 million VoIP handsets will be wireless in 2009. Figure 13 represents a block diagram of a Wireless VoIP phone [9].

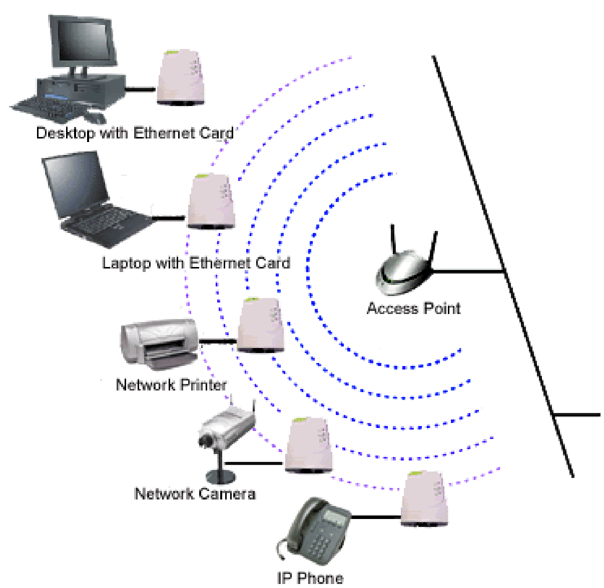

Figure 9 - Wireless Connectivity over the Internet

Combining all this technology, aren't we better off with a cordless phone that can function as a normal cell phone outside a preferred network and as a Wireless VoIP phone that can make use of that network connection to place its calls?

Wireless LAN

A wireless LAN (or WLAN, for wireless local area network, sometimes referred to as LAWN, for local area wireless network) is one in which a mobile user can connect to a local area network (LAN) through a wireless (radio) connection. The IEEE 802.11 group of standards specifies the technologies for wireless LANs. 802.11 standards use the Ethernet protocol and CSMA/CA (carrier sense multiple access with collision avoidance) for path sharing and include an encryption method, the Wired Equivalent Privacy algorithm.

CSMA/CA (Carrier Sense Multiple Access/Collision Avoidance) is a protocol for carrier transmission in 802.11 networks. Unlike CSMA/CD (Carrier Sense Multiple Access/Collision Detect) which deals with transmissions after a collision has occurred, CSMA/CA acts to prevent collisions before they happen.

In CSMA/CA, as soon as a node receives a packet that is to be sent, it checks to be sure the channel is clear (no other node is transmitting at the time). If the channel is clear, then the packet is sent. If the channel is not clear, the node waits for a randomly chosen period of time, and then checks again to see if the channel is clear. This period of time is called the back-off factor, and is counted down by a back-off counter. If the channel is clear when the back-off counter reaches zero, the node transmits the packet. If the channel is not clear when the back-off counter reaches zero, the back-off factor is set again, and the process is repeated.

All the 802.11 specifications use the Ethernet protocol and Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) for path sharing. The original modulation used in 802.11 was phase-shift keying (PSK). However, other schemes, such as complementary code keying (CCK), are used in some of the newer specifications. The newer modulation methods provide higher data speed and reduced weakness to interference.

A complementary code contains a pair of finite bit sequences of equal length, such that the number of pairs of identical elements (1 or 0) with any given separation in one sequence is equal to the number of pairs of unlike elements having the same separation in the other sequence. A network using CCK can transfer more data per unit time for a given signal bandwidth than a network using the Barker code, because CCK makes more efficient use of the bit sequences.

GSM BLOCKS

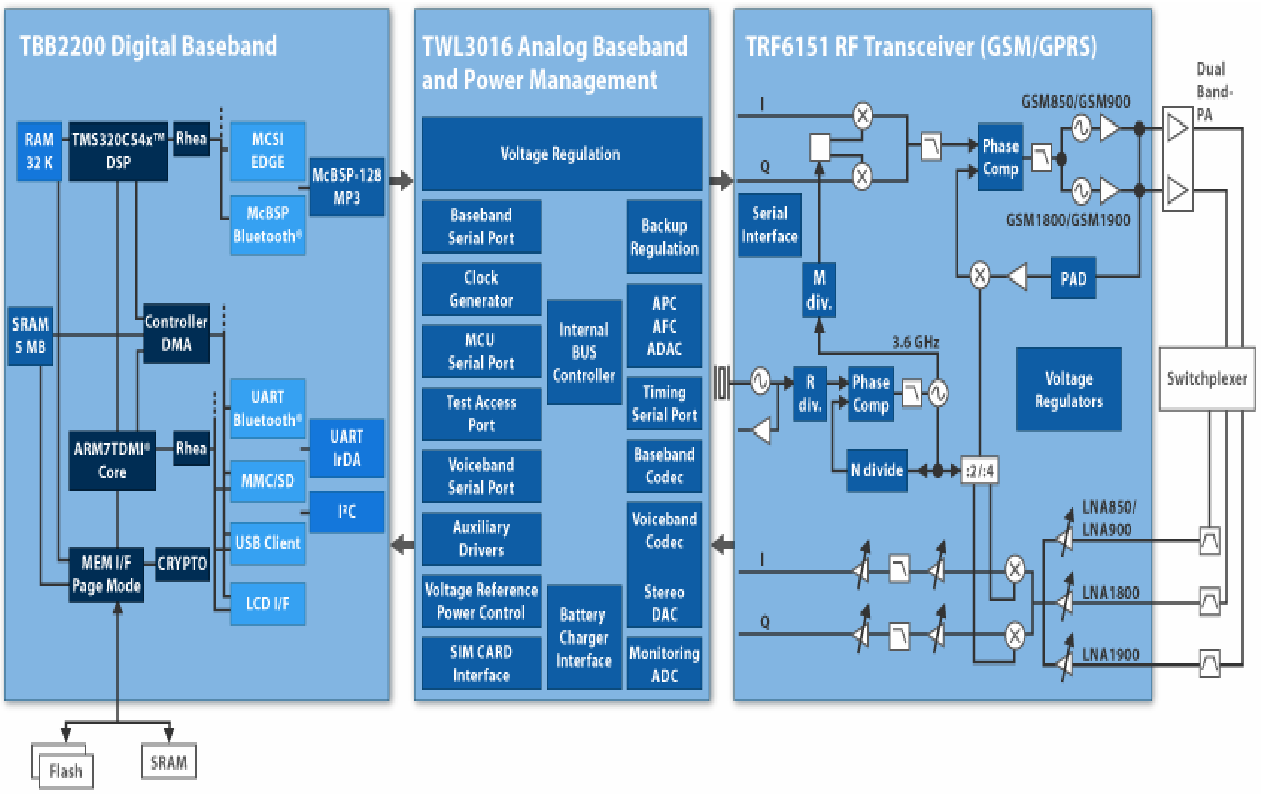

Figure 10 - Block diagram of a GSM phone

Baseband Codec

The baseband codec is a baseband digitization subsystem. Performing signal conditioning between the DSP and the IF/RF sections of the GSM telephone system [10].

It consists of an on-board ROM which contains:

1. All the code necessary for performing GMSK (Gaussian Minimum Shift Keying)
2. Two high accuracy fast DACs (Digital to Analog Converter) with output reconstruction filters

A common band gap reference feeds the ADCs and the signal DACs. The baseband functions can be accessed via the baseband serial port.

Voiceband Serial Port

This port is used for connecting digital baseband to analog baseband. It is a very invaluable component since contemporary data converters cannot be used at radio frequencies for many applications. Its function comprises of converting the received radio frequency (RF) to the intermediate frequency (IF) or baseband for digitization. Conversely, it is also used for converting the baseband or IF signal (at the output of DAC) to a radio frequency used for transmission.

Voiceband Codec

The voiceband codec consists of linear coded DACs (Digital to Analog Converter) and ADCs (Analog to Digital Converter) which maintain a wide dynamic range throughout the transfer function. They also maintain a far superior SNR (Signal to Noise Ratio) and THD (Total Harmonic Distortion) in comparison to the traditional μ-law and A-law codecs. Thus, the voiceband codec is a complete analog front-end for high performance voiceband and DSP applications. The voiceband codec can be controlled by using serial ports.

It (The Voiceband Codec) includes:

1. On-chip antialising and anti-imaging filters
2. 16-bit ADC – (three channels) completes the available auxiliary converter functions.
3. 16-bit DAC –three control DACs are included for functions as AFC, ADC and RF power control signals
4. Programmable Gain Amplifiers

Power Amplifier

The power amplifier is the biggest consumer of energy in a cell phone. It is rather a simple circuit that consists of:

1. A grounded source amplifier, which drives
2. a choke and
3. antenna

It can be driven in class A, AB or C mode.

A typical modern communications signal chain consists of transmitting and receiving sections, both of which need RF (radio frequency) power monitoring and control. A combination of the automatic gain control and power detection techniques using reference voltage set-points is commonly used to monitor the RF power in both sections.

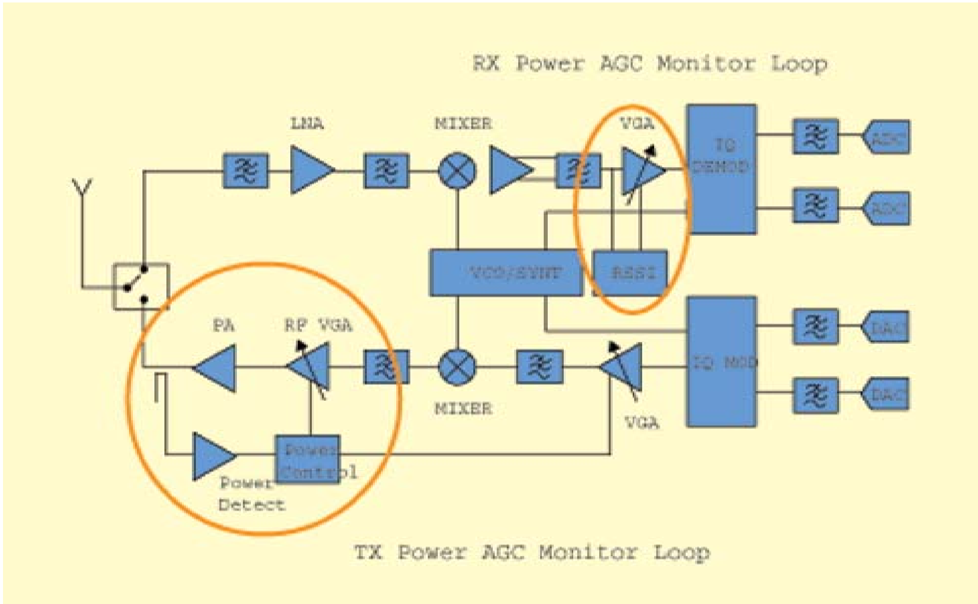

Figure 11A - Combination of power detection and automatic gain control techniques is commonly used to monitor RF power

We must be capable of processing signals of different strengths on the receiver side. This is due to the fact that changing weather conditions, or for that matter, rapid movement of source relative to the receiver, can cause the signal strength to change [11].

The output of the gain amplifier is sensed by a detector that in turn generates a DC voltage. This voltage is proportional to the RMS input of the detector.

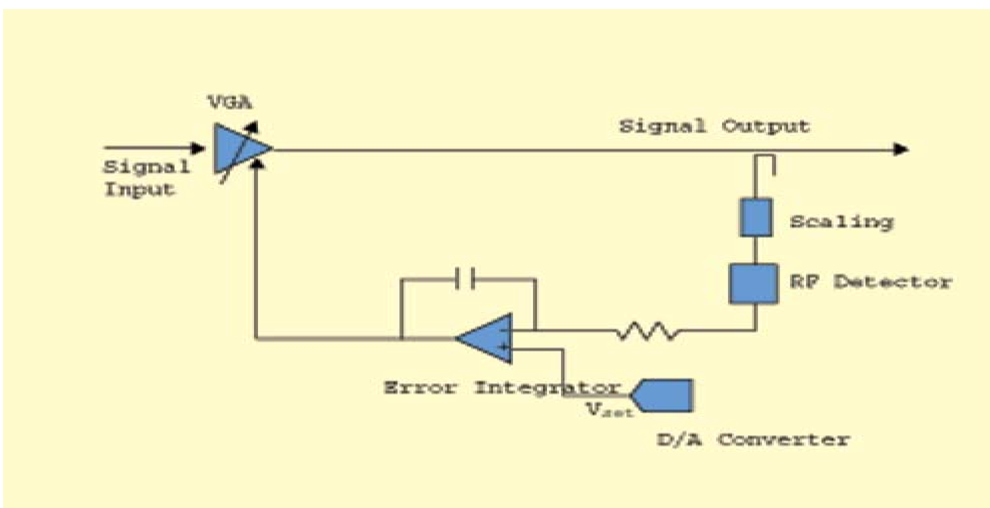

Figure 11B - The set-point voltage must be digitally programmable, so a DAC is usually used to provide the voltage

Besides performing the basic function of increasing battery life, the voltage regulators in cellular phones are used to accomplish different goals:

1. To step down the voltage between the battery and the different sub-circuits that require lower supply voltage or to step up the voltage for sub-circuits that need higher voltage than the battery (like SIM card, backlight LED, etc.). Occasionally, a buck-boost type regulator is required to generate a voltage that is between the maximum and minimum value of the battery voltage.

2. To isolate the different subsystems from each other. This is important in the RF section, and also between digital and analog/mixed signal circuits. Using LDOs can be cheaper and more area efficient than adding the traditional LC isolation filters that are used in the supply lines of RF circuits.

3. To isolate sensitive circuitry from the transient voltage changes of the battery. This is especially relevant in GSM phones where the PA operates in a 217 Hz pulsed mode with 12.5% duty cycle. The high current of the PA, typically up to 1.6A, can cause a voltage transient of up to 0.5V due to the combined effect of the battery's ESR and protection circuitry. The PSSR of the voltage regulator significantly reduces the supply transient seen by the phone circuits [12].

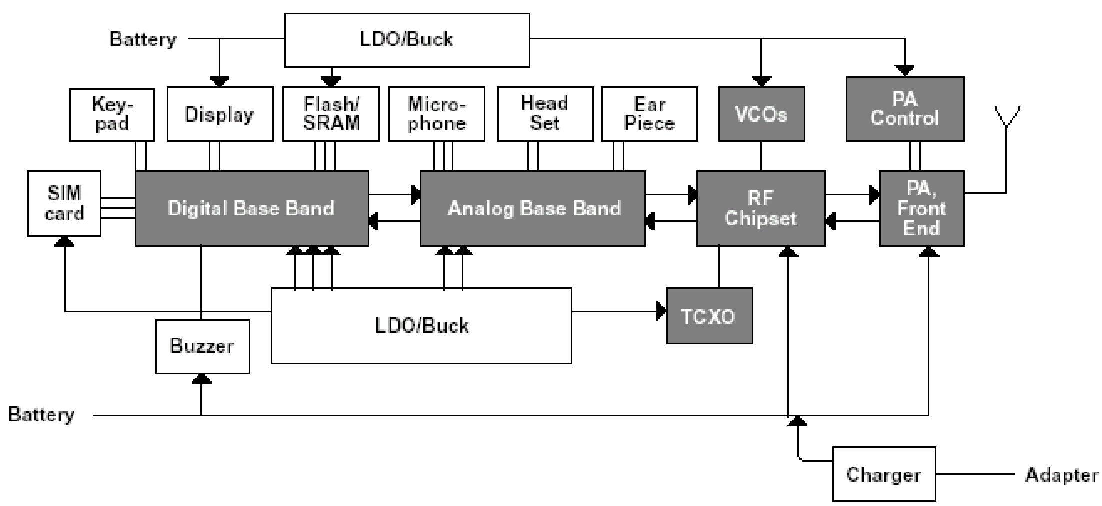

Figure12 - Typical GSM handset diagram

VoIP BLOCKS

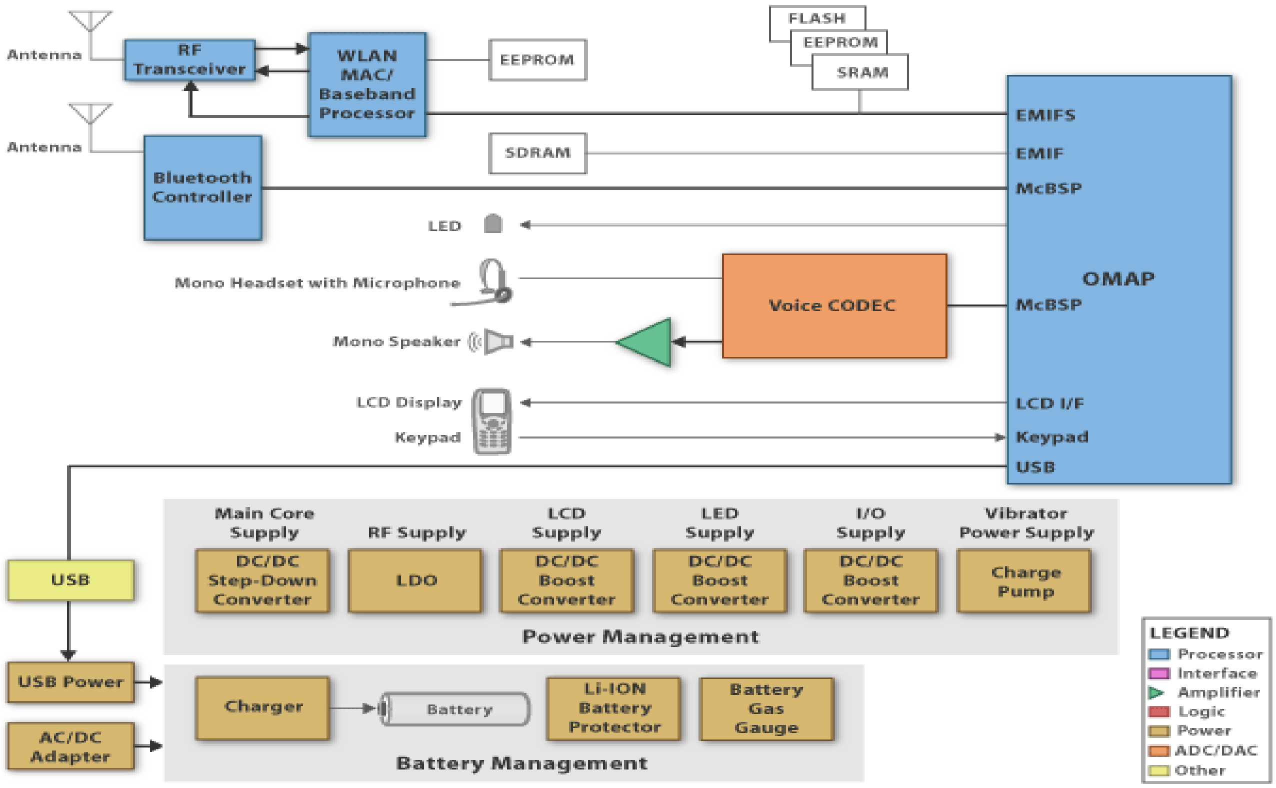

Figure 13 – Wireless IP Phone

Codec

A codec specifies the algorithms used to represent media digitally either on the wire or in storage. Various codecs are used to transmit VoIP including G.711, G.713 and G.729.

Baseband Processor (TNETW1130/ACX100)

The baseband processor provides a single chip 802.11b/g MAC/Baseband processor solution. It supports 802.11b/g and supports QoS for 802.11 after ratification. It is a complete and converged solution supporting full data transfer speeds and all 802.11 WLAN standards and draft standards. This device features seamless operation in the 2.4 GHz and 5.4 GHz bands. Additionally, the device incorporates [13]:

1. Transmit power control (TPC)
2. Dynamic Frequency Selection (DFS)
3. Other capabilities that are specified in 802.11h that are critical for deployment of WLAN

It may also directly interface to host and radio to provide a complete high-speed network solution for OEM WLAN system providers. It supports high data rates by implementing real time function in hardware.

Tnetw1130 Specifications

Key Features

- True dual-band 54 Mbps data in the 2.4-GHz and 5.2-GHz U-NII bands
- Hardware-accelerated Advanced Encryption Standard (AES) for mandatory (AES-CCMP)
- Wi-Fi Compliant 802.11g operation
- Compatible with all versions and additions to the 802.11 standard including draft standards
- Field-programmable architecture to supports standard changes
- Seamless interoperability and automatic fallback operation in the 2.4-GHz band between the higher-speed 802.11g and established 802.11b modes
- Auto-Band™ feature automatically selects between the 5.2-GHz and the 2.4-GHz bands and modulations based on user profile configurations
- ELP™ — Optimized low power consumption for maximum battery life
- Optional 802.11g+ mode provides 50% greater throughput over competitive 802.11g solutions

Dual Codec

It provides interface to the headset, microphone and speaker with on-chip drivers.

Voice Codec (PCM3500/TLV320AIC1x)

It is a 16 bit codec designed for modem analog front end (AFE) and speech processing applications. It degrades all the functions needed for a modem or voice codec, including:

1. Delta-sigma digital to analog and analog to digital converters
2. Input anti-aliasing filter
3. Digital high pass filter for DC blocking
4. Output low pass filter

It features one or more 16-bit analog to digital channels and 16-bit digital to analog channels. It provides a high resolution signal conversion from digital to analog and analog to digital using over-sampling sigma-delta technology with programmable sampling rate. It may implement the time division multiplex serial port.

Serial Interface

The synchronous serial interface provides for a simple interface to popular DSP and IRSC processors. The serial interface also supports time division multiplexing (TDM), allowing up to four codecs sharing a single four wire serial bus.

OMAP

The OMAP processor delivers the best combination of high performance and ultra-low power consumption for wireless handsets including GSM and VoIP phones. The OMAP family of solutions merges both modem and application functionality on to the existing architecture.

Digital Signal Processor Subsystem Overview

It is a collection of modules which include the TMS320C55x CPU processor along with its hardware accelerators, memory, instruction cache and DMA which it uses to communicate with the rest of the OMAP devices as well as a number of peripherals [14].

The DSP subsystem is composed of several portions:

1. The DSP module
2. The peripherals
3. The several interfaces used to communicate with the rest of OMAP modules

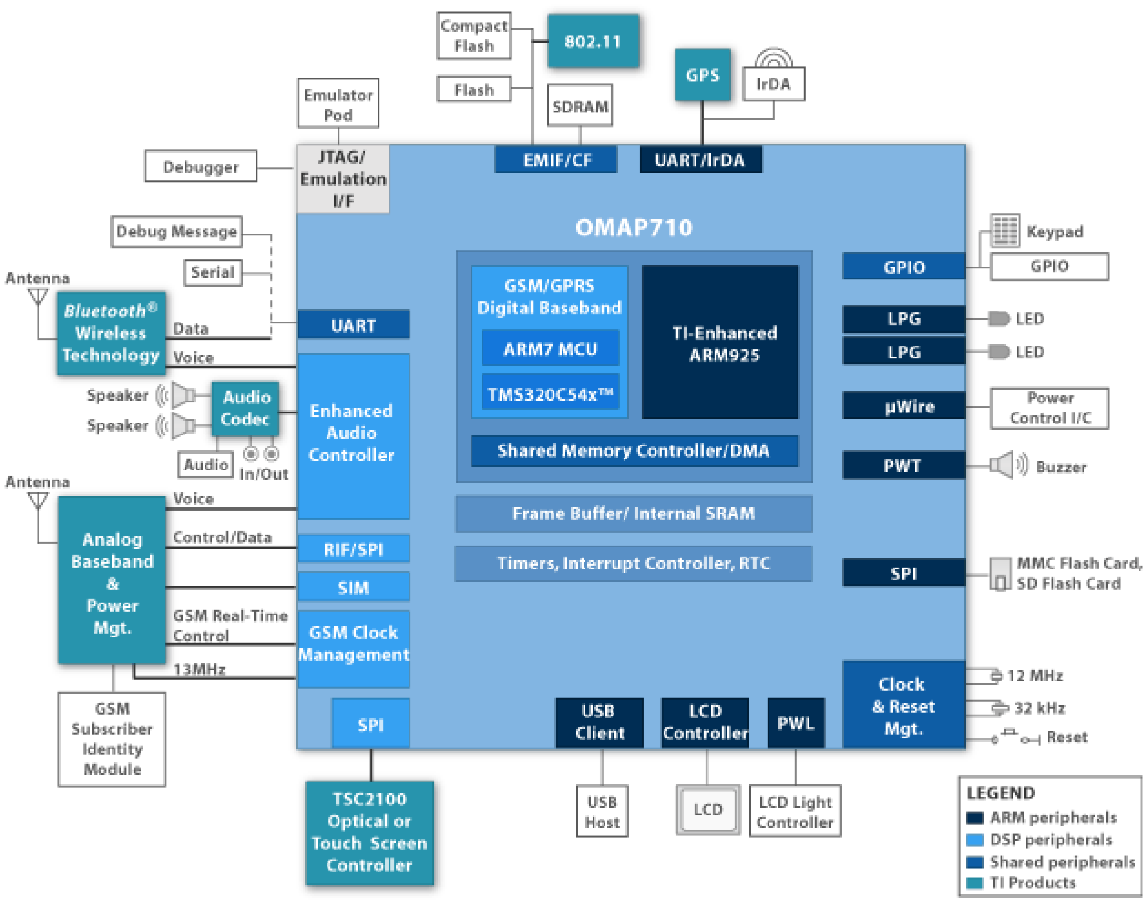

Figure 14 – OMAP710 Chip

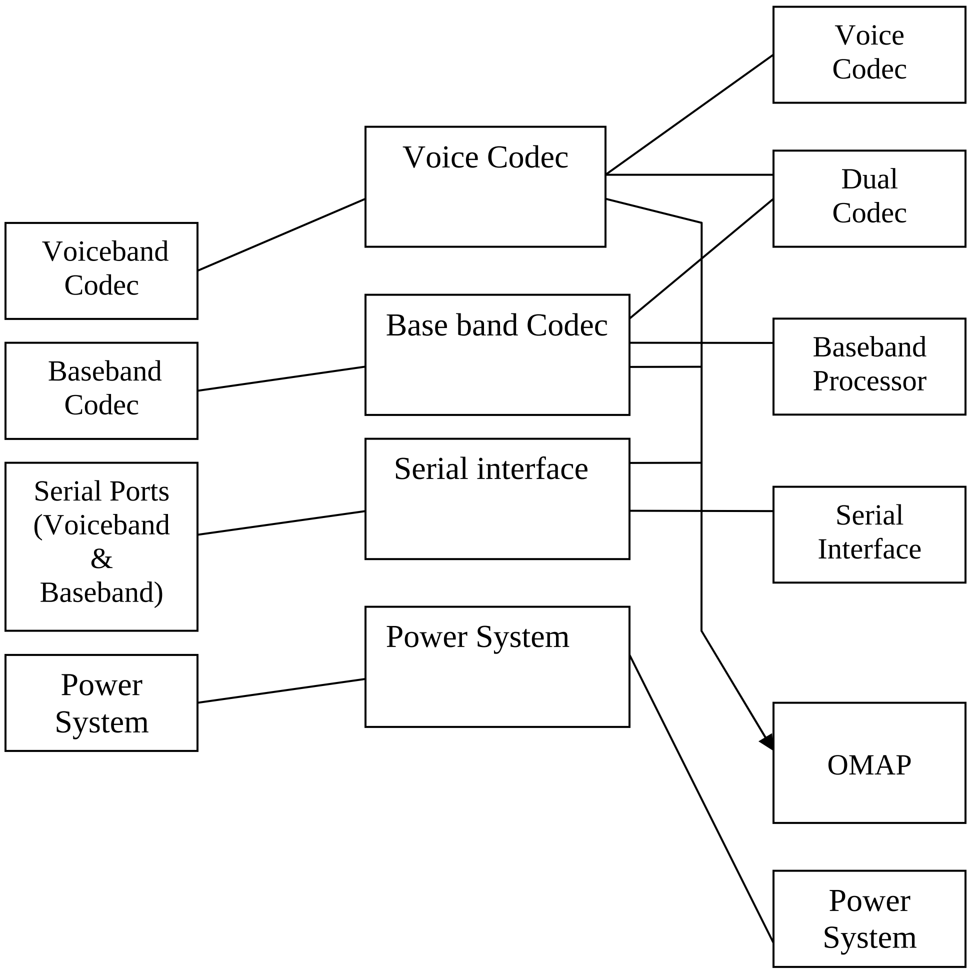

Figure 15 – Block Diagram for Dual-Functioning Interfaces

Similarly, the baseband processor, the voice codec and the serial interface perform the consequential function for the wireless VoIP phone so that it may be used by the OMAP chip (in the case of wireless VoIP phone) for its own characteristic signal processing.

Thus, we should be looking at making similar voiceband and baseband processors along with serial ports that would allow the signal to be converted into two distinct types of digital data that could be used by both of the above mentioned systems. Hence, allowing successful integration of two forms of contemporary digital communication and yet each being processed in its unique manner.

Conclusion

Arguably, accessibility has become a very prized asset of contemporary times. It weighs its value in time, thus becoming priceless. University of Hartford promises a relatively good communication network. However, is it equipped to help face with the rapidly advancing communication technology?

The popularity of an institution is judged by the opinion held of it by its beneficiaries. The chances of which are enhanced manifold if a100% accessibility to employees is provided. An improved communication setup also results in increased efficiency of an outfit.

The digital communication devices being used in the university (though being advanced) have one major drawback. All of them are landlines. Thus, many a times it is difficult to contact an individual when he's not in his office. Also, maintenance and replacement of these devices proves expensive in most cases of breakdown. This project specifically aims at removing these anomalies. Not only this, but it also provides with a technology which can be used as a base model for further upgrades necessary to help bear with future technological advancements. It lets us understand how to make a wireless communication network within the university premises. All the users may receive and make calls using their personal cellular phones, however making use of this network.

All of this is without having to build new infrastructures. The Internet WAN currently in place may be accessed by the chosen users as soon as they enter the premises, thus providing non-disruptive accessibility for all the faculty and staff. Future upgrades may include real time video communication between all the users.

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