Using Voice over IP (VoIP) in Mobile Networks
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Introduction

VoIP has been around since the mid-1990s. The International Telecommunications Union (ITU) standardized the first signaling protocol for VoIP. The protocol, classified as a conferencing standard, is referred to as H.323.

The Internet Engineering Task Force (IETF) entered the scene in the late 1990s with what became a competing protocol called the Session Initiation Protocol (SIP). In 2003 the IETF released an updated version that fixed many issues with the original protocol. Since that time, SIP has become widespread in supporting the signaling functions for most real-time applications today.

There is one other key ingredient for VoIP that involves the packaging of the bits that make up the real-time voice or video communication. The protocol that accomplishes this is called the Real-time Transport Protocol (RTP). The IETF was also responsible for developing this protocol. Apparently it never seemed necessary to develop another competing protocol to RTP, so this has been the sole standard for media transport from the beginning.

To summarize the protocols, we have these two key ingredients that are used for signaling (SIP) and transport (RTP) of real-time communications over IP packet-based networks.

One thing to bear in mind is that these same protocols are used for voice or video traffic over IP networks, which both represent time-sensitive traffic (from end-to-end). The biggest difference between voice and video IP traffic is the volume of data, or bits. Obviously, video demands a higher throughput than voice (exactly how much, depends on the resolution of the video and the compression method used). A voice or audio channel is necessary in conjunction with a video stream (i.e., silent video conferencing is not something desired by users).

The key difference between voice and video traffic versus transferring ordinary data files is the necessity to deliver a steady stream of traffic because it is time-sensitive. This is best achieved by allowing time-sensitive traffic to have priority, or right-of-way, over other types of non-time-sensitive data traffic. The term that defines this process is typically called Quality of Service (QoS).
Mobile Cellular Network Standards

There are three different groups of standards that have evolved over time that represent the majority of mobile cellular communications today.

Referencing back to 2nd generation (digital) mobile cellular networks, there were two key standards that became global standards. First, Global System for Mobile communications (GSM) was developed by the European Telecommunications Standards Institute (ETSI). This has become the most widely deployed mobile cellular voice technology in the world. The other 2nd Generation Standard is called Code Division Multiple Access (CDMA). This standard was initially developed by Qualcomm. An industry trade group was formed to provide an eco system for the cdma-One standard.

At the turn of the millennium, 2nd generation mobile standards began their evolution to 3rd generation standards. The move to 3G mobile networks would take a herculean effort on the part of engineers to accomplish the task. Therefore, regional standards bodies throughout the world formed standards partnerships in order to accomplish the task.

ETSI passed the torch to the 3rd Generation Partnership Project (3GPP), which transformed GSM into a new 3G standard called Universal Mobile Telecommunications System (UMTS) that used, what was at that time, a new air interface called Wideband CDMA, or W-CDMA.

The torch-bearer for the 3G standards development involving cdmaOne is the 3rd Generation Partnership Project 2 (3GPP2). Two 3G standards were created out of this partnership. The first became known as 1x Radio Transmission Technology (RTT) and the second was 1x Evolution Data Optimized (EV-DO). 1x stood for the original CDMA channel size of 1.25 MHz (as opposed to the W-CDMA, which was 5 MHz). The former (1xRTT) supported voice and lower-speed data traffic. The later (1xEV-DO) supported only data but at much higher data rates versus 1xRTT.

Recall that at the beginning of this section three groups of standards were mentioned. The third mobile cellular standard comes from the Institute of Electrical and Electronic Engineers (IEEE). The IEEE has created a whole series of standards involving both Local Area Networks (LANs) and Metropolitan Area Networks (MANs). These IEEE standards are formed under a group called the 802 committee, which was formed in February of 1980.

The standard that evolved into a mobile cellular standard is known as 802.16. A group of interested parties that wanted to promote 802.16 standards was formed, called Worldwide Interoperability for Microwave Access (WiMAX). That is why the 802.16 standards are also known as WiMAX. In 2004, the IEEE combined several standards documents that loosely formed the 802.16 framework into one cohesive standard, called 802.16-2004. This standard provided capability for creating a fixed-point microwave access network that provided broadband wireless connections.

The IEEE’s next move involved the migration of the fixed microwave technology into a mobile technology. The initial work for the mobile standards version of 802.16/WiMAX is called 802.16e. A newer standard called 802.16-2009 exists today that incorporates these two previous standards (802.16-2004 and 802.16e).
Note: It is necessary for the reader to have a basic understanding of the mobile wireless standards and the standards bodies that created them, since they will be referenced throughout this paper.

VoIP’s Foray into the World of Mobile Networks

The first standard for mobile networks involving VoIP was published by the 3GPP to support Fixed-Mobile Convergence (FMC). FMC was designed to extend coverage indoors where, in many cases, wireless signals are weakened as they are forced to traverse through walls.

The concept involves another IEEE standard, known as 802.11, also known as Wi-Fi. Over the past several years, basic mobile phones have given way to the smart phone. A smart phone is not only a voice terminal but a data terminal as well. Many have data apps such as browsers. Users wanting to achieve faster downloads turn on the Wi-Fi (802.11) radio in their smartphone, when in range of a Wi-Fi network, so they can achieve higher throughput for their data applications.

This is where an ingenious idea was hatched. The VoIP ingredients (SIP and RTP), previously introduced, were added to the smartphone so voice calls can be established via the Wi-Fi access point, which is connected to the Internet.

Figure 1 shows the telecommunications architecture for supporting Fixed Mobile Convergence (FMC). The key factor is supplying the user with a dual-mode handset. This means that two radios have to be embedded in the phone, a WLAN (802.11) radio along with a standard cellular radio (based on GSM/UMTS technology). The specification was initially developed by a consortium formed by mobile equipment vendors, which was dubbed Unlicensed Mobile Access (UMA). The consortium then turned the completed specification over to the 3GPP. The 3GPP incorporated it into the next UMTS standards release (Rel. 6), published in April 2005. The 3GPP refers to...
the specification within the standard as Generic Access Network (GAN). Both terms, “unlicensed” and “generic,” refer to the unlicensed spectrum that is used for Wireless LAN (WLAN), or Wi-Fi networks.

One important aspect of the standard is support for roaming between the two networks: the cellular network and the Wi-Fi, or unlicensed network. In this case, roaming refers to the ability to hand a call off between networks without the call dropping. This feature creates is a considerable challenge.

How Fixed Mobile Convergence Works

Here is a brief overview of what takes place using FMC. In this scenario, the user establishes a connection to the Internet using the Wi-Fi radio. Next, the user launches the VoIP application which then sends a registration request to the operator’s SIP Registrar server (an account in the mobile operators system and a pre-configuration must exist in the mobile device to accomplish this). Once the user device is registered, the application will indicate that calls can either be placed by dialing a number, or received when someone dials the cellular phone number associated with that account. The FMC software that makes up the VoIP application is called a softphone. Note that this VoIP application uses the SIP protocol for signaling and RTP for the media (voice) connection.

When a user enters a phone number on the keypad, or selects a phone number from a contact list, the number is relayed using the SIP protocol, along with the user device’s IP address to a SIP proxy server. The SIP proxy server interacts with the circuit-switched signaling gateway connected to the SS7 network to route the call to the appropriate destination. Once the called party answers the call, a media path is set up between the user’s device (smartphone) and a media gateway using RTP. The gateway provides a circuit-switched connection to the called party’s phone, which is connected through a circuit-switched network in this example.

As this example continues, the user decides to walk down the street to a neighbor’s house while on the call. As the user moves further away from the Wi-Fi network, used to connect the call, the signal weakens. When a certain threshold is reached by the weakening signal the smartphone requests a handover to the cellular network. This process is typically referred to as a mobile-assisted handover. The mobile device is constantly listening for broadcast messages from base stations operated by the user’s mobile operator. A list is established in the mobile device, complete with continually updated signal strength measurements. This "neighbors list" is consulted at the time a handover request is to be made to help locate best base station for a handover to be made.

Referring back to Figure 1, the Generic Access Network Controller (GANC) is the device that performs the SIP server functions and serves as the media gateway for the RTP to cellular circuit-switched connection. The GANC also forwards the handover request to the cellular network and assists in the process of alerting the mobile device of the parameters needed to switch the VoIP call from the Wi-Fi connection to a cellular base station connection that can best serve the mobile device.

Once the mobile device receives instructions via SIP messages for the call handover, the call then shifts connections from the Wi-Fi access point to the cellular base station within range for serving the mobile device. Currently, standards don’t support the ability for a call to be transferred from a cellular base station to a Wi-Fi access point via the GANC. That limitation will be removed in the next generation cellular networks, known as 4G.
4th Generation Mobile Cellular Networks are All IP-based Networks

The process previously described for FMC network to cellular network handovers is quite complex. Much of the complexity is involved in moving a connection from a packet-switched network to a circuit-switched network.

4th generation cellular networks, also known as 4G networks, don’t support circuit-switching. These networks are all IP-based. Two of the three mobile cellular standards mentioned in the introduction of this paper have formulated a standard for 4G that has been officially accepted by the ITU to meet the requirements of 4th generation networks. One standard, called LTE (Long Term Evolution)-Advanced, was developed by the 3GPP. The other standard was developed by the IEEE is called Wireless MAN-Advanced (802.16m), which is also defined as WiMAX 2.0. Neither of these standards is ready to be implemented at the time of this writing (April, 2011).

However, the precursor to these standards; LTE and 802.16e/802.16-2009 are both all-packet-based networks as well, even though they don’t have all the ingredients to map up to the rigid requirements set forth by the ITU’s International Mobile Telecommunications (IMT)-Advanced standard. This is an important footnote because “4G” is the new marketing term used to define various standards improvements made to existing networks, even if they don’t meet the ITU’s IMT-Advanced requirements.

An important point to be made regarding these newer emerging technology implementations is that there are plenty of advancements to warrant significant fanfare. With that said, the remainder of this paper will focus on how VoIP implementations are deployed for LTE and WiMAX networks.

Introduction to LTE

We now turn our attention to the emerging 4th generation technology, called Long Term Evolution (LTE), that was developed by the 3GPP. To better understand where LTE fits into the 3GPP standards path, it is necessary to provide a short explanation on the various releases.

The 3GPP assumed control of the 3rd generation standards development for GSM from ETSI in the 2000/2001 time-frame. ETSI continues to be involved in the standards development process; however, there role has changed to become one of five regional standards bodies involved in the development of the Universal Mobile Telecommunications System (UMTS) standard, which was referred to as release 4 (Rel. 4).

UMTS releases 5, 6, and 7 involved significant modifications to the W-CDMA interface to improve performance. A specific technology upgrade was introduced to the 3G standard called High Speed Packet Access (HSPA). The later, Rel. 7, is called HSPA+ or HSPA Evolved, meaning that it was to be the last release for UMTS standards that uses the CDMA air interface.

Release 8 is called LTE. There are several reasons why LTE Rel. 8 is referred to as a 4th generation mobile wireless standard. First, a significant change was made in the air interface (wireless connection between the user equipment and base station) from CDMA to Orthogonal Frequency Division Multiple Access (OFDMA). Suf-
Voice over LTE

From the information in the previous section, the reader now knows that a mobile operator’s move to LTE causes a migration to an all-IP network. This is a major step for mobile operators, and it poses some challenges. The majority of revenue generated by mobile operators is from voice and texting services. Data also generates solid revenues, but uses a disproportionate amount of bandwidth for the volume of revenue that is generated for the service.

Therefore, mobile operators need to continue offering a quality voice service as they transition to an LTE networks. The industry has identified three options that carriers can deploy to provide voice services for LTE networks.

IMS for Voice over LTE

LTE core networks are designed to work with the IP Multimedia Subsystem (IMS). The IMS was also designed by the 3GPP. It is an architectural framework for delivering and supporting IP multimedia services. Using the IMS to provide voice services over an LTE network makes perfect sense. However, many carriers do not have all the IMS pieces in place yet. Therefore, it is the best choice of the three options presented here, but it is viewed as the mid-to-long-term goal.

Since the IMS provides support for multimedia, it is only logical to consider what it can provide in the way of multimedia services that go beyond voice. Video teleconferencing can also be supported by the IMS. Notice a current trend with newer smartphones that are equipped with front and rear facing cameras. The front facing camera is designed to provide the hardware capability for software applications designed to support video calls.

The IMS uses the SIP protocol to support all of the signaling services. This is a significant point to make in terms of why SIP eclipsed H.323 as the preferred protocol for VoIP signaling. The 3GPP standards represent nearly five billion subscribers worldwide. It stands to reason that when the 3GPP selected SIP as the signaling protocol for the IMS that the industry took notice. The author believes wholeheartedly that this is the primary reason why SIP, and also the IMS, is being adopted by all telecommunications carriers worldwide.

It should now be clear that IMS provides a VoIP solution for voice over LTE. RTP is used for the media, whether making a voice call or a video teleconferencing call. When all the pieces are in place for IMS by mobile operators and carriers alike, this will make for a cohesive platform allowing a smooth interoperation between networks, both wired and wireless, for VoIP communications.
Circuit-Switched Fall Back (CSFB)

CSFB was adopted by the 3GPP as the preferred method for supporting voice services over LTE networks in the short term. The term "fall back" means that the LTE user equipment reverts back to using the mobile operators 3G network for voice calls.

There are challenges with CSFB. This solution requires that the user equipment must be dual-mode, having both circuit-switched 3G functionality along with packet-switched LTE functionality built into the device. This does allow the mobile operator to preserve the investment in their existing voice network resources, but creates a rather clunky solution for the user.

Whenever the user makes or receives a call, the user equipment must disconnect from the LTE network and connect to the operators 3G network that handles the circuit-switched voice call. When the user completes the circuit-switched voice call, the user equipment disconnects from the 3G network and re-establishes a connection with the 4G network. This means the user cannot simultaneously use the LTE network to surf the web while on a voice call; only one connection at a time is supported. This CSFB option doesn’t have any affiliation with VoIP.

Voice over LTE via Generic Access (VoLGA)

Several mobile equipment manufacturers disagreed with the 3GPP that CSFB is the best short-term solution. In 2009, several of these manufacturers banded together to form the VoLGA Forum to champion an idea that they felt was a more elegant solution to the voice over LTE issue.

VoLGA has a lot in common with FMC, covered earlier in this paper. The basic idea is simple. The user initiates a call using an LTE data connection linked to a gateway in the LTE core network. The gateway is called a VoLGA Access Network Controller (VANC), used to establish the call connectivity between the LTE packet network connection and either a GSM or UMTS mobile circuit switch. Quality of Service (QoS) can be enforced to provide high-quality voice calls because all the equipment interfaces and connection are in the mobile operator’s control.

An interesting observation about VoLGA is that it supports both circuit-switched and IMS-based services. Studies conducted by Deutsche Telekom realized a call-setup time was improved nearly 40%. The VoLGA Forum has completed the development of the specifications designed to deliver voice and SMS services for users as mobile operators transition from GSM/UMTS and LTE networks. In spite of these statistics, support seems to be waning for VoLGA.

There are some mobile operators, like Verizon Wireless, involved in a more challenging transition, from 1xRTT/1xEV-DO networks to LTE. The 3GPP2 must get involved to support protocols that bridge network functionality between 1xRTT/1xEV-DO and LTE networks.

Introduction to WiMAX

WiMAX is fundamentally similar to LTE in that it is also an all-IP-based network. The air interface between the user device, called a Subscriber Station (SS) in WiMAX networks and the base station also uses OFDMA, just
like LTE. The details of how these networks are implemented with regard to the nodes used, interfaces between
them and protocol structure is different.

LTE has the advantage as an outgrowth from the most widely deployed mobile technologies throughout the
world. The subscriber base is almost five billion strong (including legacy standards going back to GSM). The
large subscriber base creates an environment for economies of scale in manufacturing, making a huge impact to
reducing the cost of products, that pertains to both the infrastructure side and user equipment.

Mobile WiMAX does not have this advantage. Simply stated, it is in its infancy when compared to other mobile
wireless standards that have been deployed worldwide for decades. However, not having to deal with legacy
networks has its advantages. Recall from the previous section the dilemma of the best way to provide voice over
LTE, for the short term, and also for the long term.

WiMAX doesn’t have the same concerns with forward/backward compatibility issues with a subscriber base.
Various WiMAX networks have been trying out different vendor implementations of VoIP since mid-2000. This
dates back to the installed WiMAX base prior to mobile WiMAX, when systems were fixed location systems.

It is true that WiMAX network operators that deployed fixed-based systems and then adopted the mobile
WiMAX network architecture faced difficulty with upgrading subscribers because the two standards (802.16-
2004 and 802.16e) were not compatible. That problem has been largely resolved with the adoption of the air
interface technology used in the mobile WiMAX standards. Since the WiMAX operators had relatively small sub-
scriber bases, when compared with other mobile cellular operators, the painful period of switching to the newer
technology was somewhat mitigated.

Since WiMAX has an all-IP core network, supporting VoIP is very straight forward. There are no legacy issues to
deal with, other than the one between fixed and mobile that was identified above. The only real challenge is to
identify in the details the best way to roll out VoIP for the mobile devices in an agnostic way that won’t involve
proprietary solutions. That way all WiMAX operators can choose between different vendors’ equipment solutions
that will plug and play with one another vendor.

The biggest WiMAX operator in the world is Clearwire, in the United States. The company now has over four mil-
lion subscribers in 71 markets. WiMAX network operators, like Clearwire, have one advantage over LTE in that
the standards were available to begin deployments back in 2007/2008, after the first round of interoperability
testing was completed, and equipment was certified by the WiMAX Forum for Release 1.

In contrast, Verizon Wireless launched their LTE network in December 2010. WiMAX network operators had an
approximate advantage of time-to-market of roughly 30 months over LTE. However, that lead is quickly evapo-
rating, due to the fact that Verizon Wireless now covers 39 markets.
Voice over WiMAX

Fixed WiMAX services were designed to compete with wired broadband services. It was difficult for WiMAX to compete with the bandwidth available on cable networks. However, it is much easier for them to compete against Digital Subscriber Line (DSL) broadband services that are offered by traditional telephone companies.

The key component to higher adoption rates for broadband services is in the packaging. The term “triple play” is used to define a service package many are very familiar with: voice, broadcast video (and usually some form of on-demand video), and high-speed data access to the Internet. Average Revenue Per User (ARPU) is an important number for mobile operators. WiMAX operators can increase their ARPU by implementing VoIP services, along with data services.

As newer standards of WiMAX are implemented, throughput will continually rise. This provides the opportunity to offer video broadcasts, in addition to voice and Internet data services, that allow them to compete in the triple-play market. But there is more. Since WiMAX is mobile, it allows all these services to become portable. When you add mobile service to a triple-play package, it becomes what is now referred to as a quad-play.

The Future is Bright for VoIP in Packaged Services

In many broadband markets, cost-conscious subscribers opt out of paying for triple play when they have good cellular coverage at their residence. Many subscribers are willing to forgo the landline because their cell phone has become their primary means of voice communication, and they don’t want to pay for a service rarely used.

In the future, quad-play will become the new norm for two reasons. First, carriers/mobile operators are expanding their service offerings to become a single solution service provider. Second, people want portability of the services that they are paying for, and 4th generation mobile networks are a way to accomplish that.

Since the trend of moving away from traditional land-line voice services in favor of mobile voice service began, the cost-effective way to provide this service is with VoIP. All-IP networks are the future for all service providers. VoIP is a natural application extension of an all-IP network core.

Over-the-TOP

VoIP can be viewed as a disruptive technology. Early in the last decade, VoIP was viewed by service providers as a threat to their traditional revenue streams. This has turned out to be true. Today, Vonage service allows their customers to call over 60 countries for a flat rate that is minimal compared to previously high-priced international long-distance charges. This is all made possible by VoIP and the Internet, which is NOT distance-sensitive with respect to the cost of access. The only necessary ingredient for the subscriber is a broadband connection.

Over the past several years, mobile operators have been providing mobile broadband for their subscribers. These mobile operators face the same issue with VoIP; a disruptive technology that threatens their traditional revenue stream, which are subscriber fees for voice and texting.
Carriers simply need to adopt VoIP as their primary means for voice service offerings; otherwise, the subscriber will go elsewhere. The previous statement is not to imply that the subscriber will abandon their mobile operator for another, they will simply go around the problem of higher-cost voice services by downloading and installing third-party VoIP applications, such as Truphone, Bria, Skype, etc.

If the mobile operator delays getting VoIP services rolled out quickly to their subscribers, their subscribers will use applications that don’t cost them any extra to use, especially with unlimited data plans. Using third-party VoIP applications on a mobile device is referred to as over-the-top service. These services are based on a best-effort Internet connection (no QoS), but at little to no cost of use. This is a compelling argument for putting up with some dropped packets.

**Conclusion**

This paper has identified the connections with VoIP in mobile networks. It is clear that VoIP will play an even more prominent role in all telecommunications networks moving forward. It will one day be used completely in place of circuit-switched voice.

An important feature with VoIP is that it is so flexible because of the IP networks it runs on. Because different codecs (analog voice to digital converters) can be incorporated into user devices and SIP can be used to negotiate a given codec for a call, this will allow significant improvements in voice quality. Before this decade comes to a close, the author predicts that the majority of the population will experience high-definition voice on the majority of phone calls made. This will truly be an experience to behold!

**Footnote**

1 Data obtained from an article titled Clearwire Losses Widen; Buildout, Phones Stalled in Wireless Week dated: February 18, 2011

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