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Current Challenges and Opportunities in VoIP over Wireless Networks

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1. Introduction

The tremendous emergence of different wireless technologies has created many opportunities for extending many Internet applications and services to the wireless domain, and has envisioned many new ones. When we refer to wireless networks, people thoughts go directly to the Wireless Fidelity (WiFi) technology. However, WiFi networks are one type of the wireless networks that have been used for several years on a small scale areas and buildings such as schools, offices, companies, restaurants, and airports. The other types of wireless networks that cover larger geographical areas are the third and fourth Generation wireless networks, abbreviated as 3G, 4G networks, respectively, which basically use the mobile infrastructure. Some examples of these networks are High Speed Packet Access (HSPA), Worldwide Interoperability for Microwave Access (WiMax), and Long Term Evolution (LTE). In this book chapter, when we refer to the wireless network, we refer to both types of networks unless we specify one of them.

In the future, people are expected to rely more on wireless networks. This is due to the fact that these networks are easy to deploy and manage, support mobility, and provide fast Internet service, especially with the latest HSPA+ and LTE technologies which can provide relatively high speed up to 30 Mbps. As such, many Internet applications and services are used by people over the wireless domain, which may impose more challenges that did not appear in the wired networks. This is due to the natural of the wireless networks that is more prone to interferences and noises that are generated by the transmission media and atmosphere. In other words, transmitting data over wireless networks result in more packet losses than the wired networks, which will in turn cause more retransmission attempts in case of Transmission Control Protocol (TCP) based applications and traffic, such as Hypertext Transfer Protocol (HTTP) traffic and more packet loss and performance degradation in case of User Datagram Protocol (UDP) based applications and traffic such as multimedia traffic. As such, transmitting over wireless networks needs more mechanisms and methods to detect and possibly correct these network impairments that did not exist under the wired environment.

Transmitting voice over the Internet protocol or what is widely used as Voice over Internet Protocol (VoIP) is one of the promising applications that proved its popularity in the past decade and started to be used by people over the wireless networks, especially with the

advent of the smart mobile phones that are capable of supporting many VoIP protocols such as the Session Initiation Protocol (SIP), which opens the horizon to the developers and programmers to write mobile applications (apps) that can conduct VoIP calls over the mobile phones. These apps gain popularity by mobile users since they not only save them a lot of money especially when conducting international calls, but also provide them the mobility and freedom they used to have using their mobile phones. However, in order to eliminate the need of typical mobile phones and provide a good substitution for the typical phones, VoIP services have to be equipped with mechanisms and algorithms to guarantee some Quality of Service (QoS) assurance especially when it comes to packet losses which is the most annoying network impairment factor to the VoIP users. VoIP users are aware that a superior toll quality service cannot be achieved over the Internet, due to the well known fact that the Internet is not well suited to provide real-time services, and also to the fact that users pay much less money to the VoIP calls, when compared to the expensive circuit switched phone calls, so they can tolerate some quality degradation. However, they still expect an acceptable quality that does the job and will not end up wasting their time and money.

Many VoIP vendors have realized the future trend toward the wireless and mobile environments and started developing customized VoIP apps that can be installed on the smart mobile phones. Skype is a very well-known example of a successful and widely used VoIP application that is used by millions of people over the entire world. Skype has recently issued a mobile app that allows users to conduct VoIP calls over their mobile phones. However, the developed app does not take into account all the aspects and challenges imposed by the wireless domain. Therefore, the call performance of the mobile app was much lower than the wired app. This fact was validated by [1] who has done a performance profiling study for Skype over different wireless networks and environments.

In this book chapter, we aim to highlight some of the challenges introduced by different wireless technologies, and how they affected the performance of some Internet applications that migrated directly to the wireless domain without taking into account the differences between the wired and the wireless networks. VoIP will be the main application of interest in this book chapter. However, similar algorithms and schemes can be developed for other applications such as audio and video streaming over wireless networks.

The structure of the chapter is divided as follows: Section 2 will provide a quick overview about VoIP service in general followed by a more detailed description about the typical wireless network infrastructure used for carrying VoIP calls. Section 3 will introduce some of the common used VoIP mobile applications available in the market and will focus on the main challenges introduced by the wireless media and how they affect VoIP performance. It will also provide a performance evaluation profiling for one VoIP application over the wireless network. Section 4 will shed the light on some of the state-of-the-art solutions and frameworks proposed in the literature and how they addressed some of the afro mentioned challenges, Finally, Section 5 will conclude the chapter.

2. VoIP network infrastructure

We first open this section by defining what is meant by VoIP and the basic building blocks for providing such a service. As shown by figure 1, carrying voice over the IP network or

VoIP is achieved by first converting the voice signal into the digital domain and this can be done using Analog to Digital Converters (ADC), which are embedded into the computer sound card or installed inside the mobile phone audio signal processing unit. Once the voice is converted into the digital format, it can be packetized into IP packets that are ready to be transmitted utilizing the UDP transport protocol. UDP is preferred over TCP due to the fact that UDP is much faster, simpler, and does not impose acknowledging the reception of each packet, which saves a lot of time and makes it more appropriate for real-time applications such as VoIP. Once the VoIP packets are transmitted over the transmission media, some packets may get lost, and at the receiver side, the received packets are de-packetized into a stream of ones and zeros. After that, the binary stream is fed to the Digital to Analog Converter (DAC) unit which converts the digital bit stream back to the analog format and play it back.

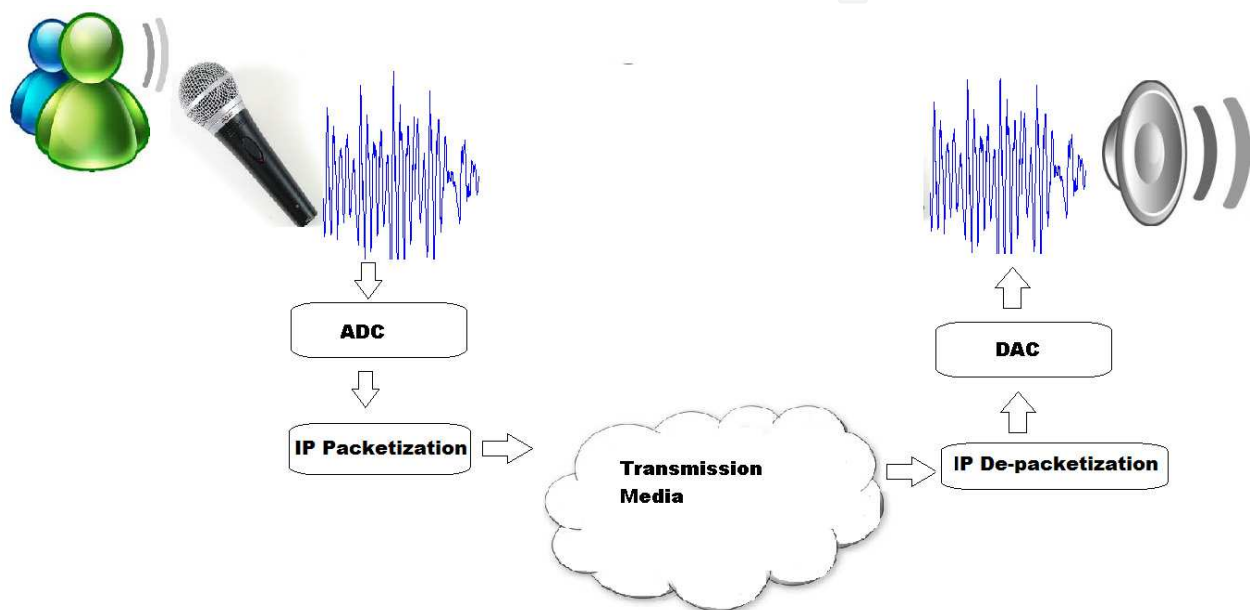


Fig. 1. VoIP packetization and transmission

Now, in order to transmit the sequence of packets over the Internet, some networking protocols and services have to be deployed. Figure 2 depicts the network architecture for a VoIP enabled network, including both wired and wireless VoIP terminals, utilizing WiFi wireless network infrastructure. For simplicity, we assume that the Session Initiating Protocol (SIP) is the protocol used for signaling purposes and call set up. Considering the fact that the main focus of this section is to familiarize the reader with the major components and the architecture of the VoIP network, we refer the reader to look at [1, 2] for more hardware and software details.

The following scenario shows how a VoIP call over wireless network is conducted. Let us suppose that user A in Domain 1 would like to call user B in Domain 2. When user A accesses the wireless network using his VoIP enabled mobile phone, the Quality of Service Access Point (QoSAP) uses Call Admission Control (CAC) algorithm to determine whether or not the wireless network can safely admit the new VoIP call. Different access points may deploy different CAC algorithms, but the common factor for all these different algorithms is to accept/reject calls based on the wireless network capacity. If the call is admitted, user A will send a call setup request to the VoIP softswitch (SIP INVITE message) shown in figure

3. The called softswitch (Proxy server in case SIP protocol is used or Gatekeeper in case H.323 protocol is used) [3] will perform some operations; such as authorization, authentication, and accounting. After that, the softswitch will lookup the IP address of the destination domain (Domain 2). Notice that in some networks, address lookup can be done in the local Domain Name Server (DNS) of the network. Once the destination IP address is determined; the VoIP gateway checks the link status by using an appropriate CAC mechanism. End-to-End Measurement-Based Admission Control is one of the widely used admission control mechanisms in VoIP gateways [4]. The VoIP gateway gauges the quality of the network path by sending probes to the destination IP address, which is usually the IP address of the destination gateway or the destination softswitch, and measures the end-to-end delay, packet loss, and jitter delay of these probed packets to determine the quality of the network path [4], which will be reported to the softswitch. The VoIP gateway uses these information to determine whether the call can be admitted or not. If the call is not admitted, the connection is rejected and a busy signal will be sent to the VoIP terminal. The process is repeated till the call is successfully established.

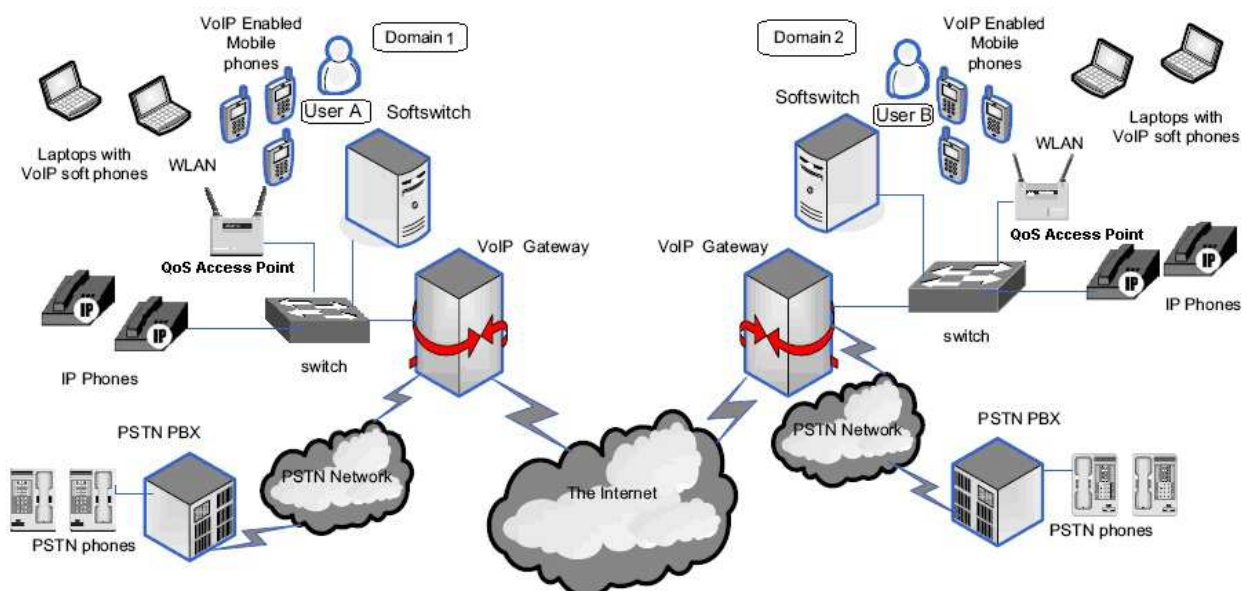


Fig. 2. Network architecture for VoIP over wireless networks [12]

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INVITE SIP: USER A @ Domanin1.com SIP/ 2.0
Via:SIP/2.0/UDP
From: sip: USER A @ Domain1.com
To: sip: USER B@ Domain2.com
:

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Fig. 3. SIP invite message format [12]

Notice that the above scenario assumed the usage of QAP that deployed some CAC algorithm to accept or reject phone calls. In addition, another CAC algorithm is deployed at the networking level to determine the quality of the end-to-end Internet connection and whether it meets the minimum VoIP quality requirements. In other words, one should take

into account that neither the QAP nor the CAC algorithm may exist in case of wireless VoIP, which may add extra challenges for providing an acceptable VoIP service.

Figure 4 shows another possible scenario in which a 3G/4G wireless Internet connection is used. In this case, the challenge of deploying VoIP is even more due to the fact that the wireless link covers larger distance ranges to hundreds of meters when compared to the wireless LAN.

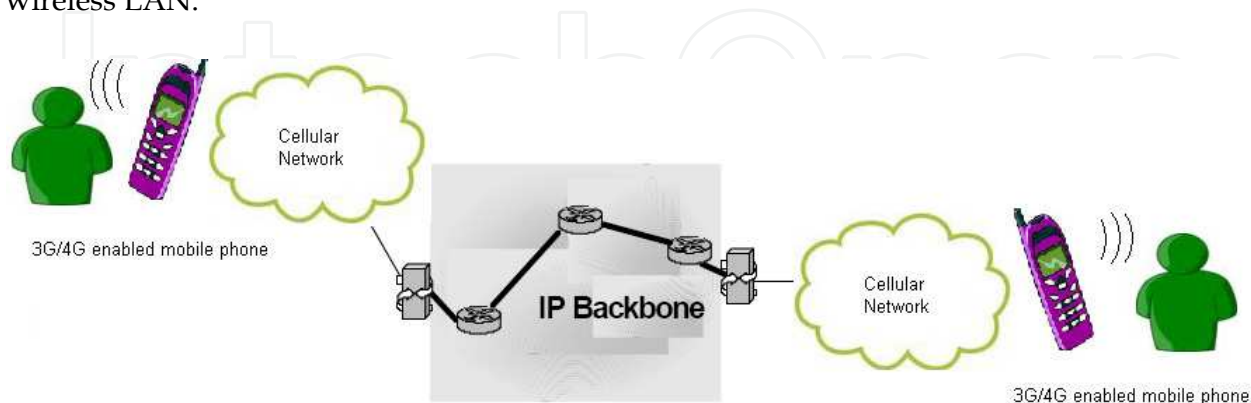


Fig. 4. 3G/4G wireless VoIP call path

3. Current VoIP applications, performance profiling, and challenges

Many VoIP vendors and developers realized the tremendous emergence of different wireless technologies and also the widespread adoption of smartphones. In fact, according to the Disruptive Analysis research firm [5] which expects that the number of VoIP-over-3G users will be around 250 million by the end of 2012. Accordingly, many of them start developing mobile applications that are capable of conducting VoIP call utilizing the wireless Internet connection supported by the mobile phones. In this section, we will explore some of the most famous VoIP mobile applications and their supported features. We will then focus on one of these applications, Skype particularly, and investigate its performance over different wireless networks.

3.1 VoIP applications

In what follows, we briefly describe some of the current widely used VoIP applications specifically customized for mobile phones and terminals. As an example, we will provide a brief description of the following applications: Truphone, Fring, and Skype.

3.1.1 Truphone

Truphone is considered as one of the pioneers in developing mobile apps that is capable of conducting VoIP over the mobile's wireless internet connectivity [6]. The company even provides its customers a VoIP number that can be reached from the Internet, which may be useful especially when there is no or weak network coverage from the mobile provider. In addition, calling this VoIP number will be free of charge for the people who call from/to the Internet. Furthermore, they have developed a new service called Call Through, in which Truphone customers can still conduct a low cost VoIP calls even if they do not have a WiFi or 3G Internet connection on their mobile phones. This works by first conducting a local

mobile call to a Truphone access number, which conduct a VoIP call to the final destination. In this way, the user will be charged for the local mobile phone call and the cheap VoIP international call.

3.1.2 Fring

Another well-established company that offers several services such as video, voice calling, and instant chat messaging all utilizing the Internet connectivity of the mobile phone [7]. As shown in figure 5, the application provides the users an interface similar to the typical mobile interface, which makes it both easy and attractive to the normal users. Furthermore, Fring users can enjoy other services that are not available in the normal mobile phones such as video conferencing, communicating with the Internet users using instant and text messages, and calling Skype users.



Fig. 5. A screenshot of the Fring VoIP application on smartphones

3.1.3 Skype

With over 600 million users, Skype with no doubt is the biggest and largest VoIP service provider that shapes and leads the industry in this field [8]. As any other provider, Skype decided to provide its services over mobile phones by developing a mobile app for the current smartphones. However, as will be shown in the next section, the performance of Skype over the wireless network is much lower than the wired network, and that is due to the fact that most of the current developed applications do not take into account the characteristics of the wireless channel and devices. As such, migrating the application designed for the wired network needs to take into account the additional challenges and difficulties introduced by the wireless channel.

3.2 Performance profiling of skype over wireless networks

To demonstrate the fact that the current VoIP applications designed for the wired environment need to be further developed and enhanced such that it does not degrade under the wireless environment, we will briefly describe the research work of [1] which

experimentally shows that the performance of Skype degrades dramatically under different wireless environments. In what follows, we explain to the extent level the conducted experiments used in the performance evaluation, the performance evaluation results, as well a summary of what are the necessary enhancements for improving the quality of voice after being transmitted over mobile environments.

3.2.1 Scenario setup

All experiments are conducted utilizing Experimental Testbed for Research Enabling Mobility Enhancements (EXTREME) [9], which is developed for testing network algorithms and technologies of the Centre Tecnològic de Telecomunicacions de Catalunya (CTTC) in Barcelona, where the Wireless Local Area Network (WLAN) infrastructure is supported and its configuration is made with Asymmetric Digital Subscriber Line (ADSL) of rate 1 Mbps and 300 Kbps for uplink and downlink, respectively. Moreover, the production network operator of Orange mobile is directly interconnected with it via loosely coupled mechanism. Skype clients are used as either PC (Pentium IV with 512 MB RAM memories) or mobile terminal (SPV M5000 or Otek 9000) nodes. The SPV M5000 uses a processor of Intel Bulverde 520 MHz with 64 MB cache memory. The Access point used in WLAN uses Atheros-based cards integrated with Madwifi driver [10]. The scenario setup used is depicted in figure 6.

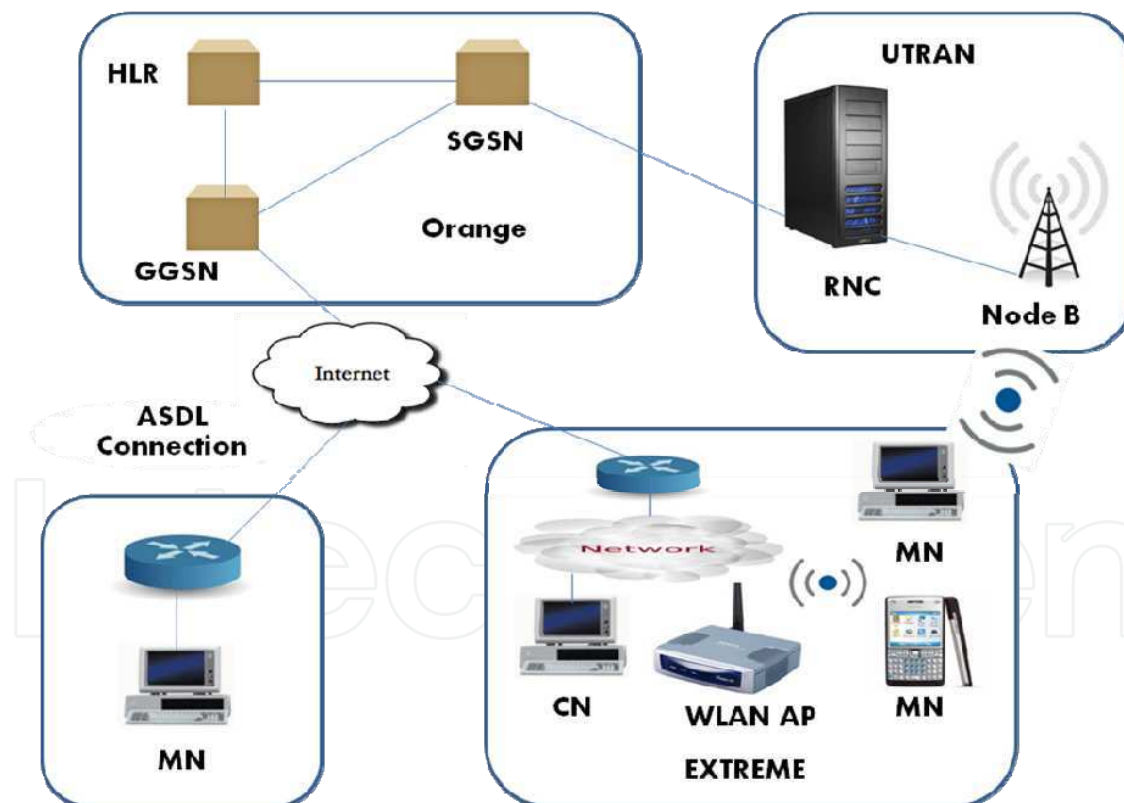


Fig. 6. Scenario setup [1]

3.2.2 VoIP quality measurement and evaluation

To measure the voice quality, the Perceptual Evaluation of Speech Quality (PESQ) algorithm, which is described by the ITU-T recommendation P.862, is used [11]. In this

algorithm, a comparison is made between the degraded analog audio signal, which is the signal transmitted over the communication channel, with the analog audio reference (original) signal. Consequently, the call quality is evaluated using the Mean Opinion Score (MOS) performance metric which represents the average quality score considering a wide set of subjects. The range of MOS usually varies between -0.5 to denote for poor perceived voice quality and 4.5 to describe the best possible obtained voice quality.

About experiments setup, there are two clients (Client 1 and Client 2) considered, as shown in figure 7, where the VoIP session is established between them. Furthermore, there are also two PCs, PC 1 and PC 2 connected with Client 1 and Client 2, respectively. Client 1 represents a PC-based node while Client 2 indicates either a PC-based node or SPV M5000 Personal Digital Assistant (PDA). In all experiments, the adopted network of Client 1 connection is LAN while this is not the case for Client 2 connection that depends on the scenario being under test. During the voice session between Client 1 and Client 2, a WAV file which shows a spoken English text without noise is recorded by PC 2 using the *wavrec* application and consequently reproduced by PC 1 using the *wavplay* application. Therefore, two audio cables are required in all tests. The first one connects the microphone jack of Client 1 with earphone jack of PC 1 and the vice versa for the other one considering PC 2 and Client 2. The process of recording at PC 2 and reproducing at PC 1 occur simultaneously using Network Time Protocol (NTP) and due to end-to-end delay, it requires a minimum duration of 20 seconds including silence at the beginning and the end of WAV file. Also, it is automated using a script that is also responsible for estimating the PESQ score between the resulting WAV files. On the other hand, the WAV file is sampled at 8 KHz sampling rate and encoded with 16 bits per sample.

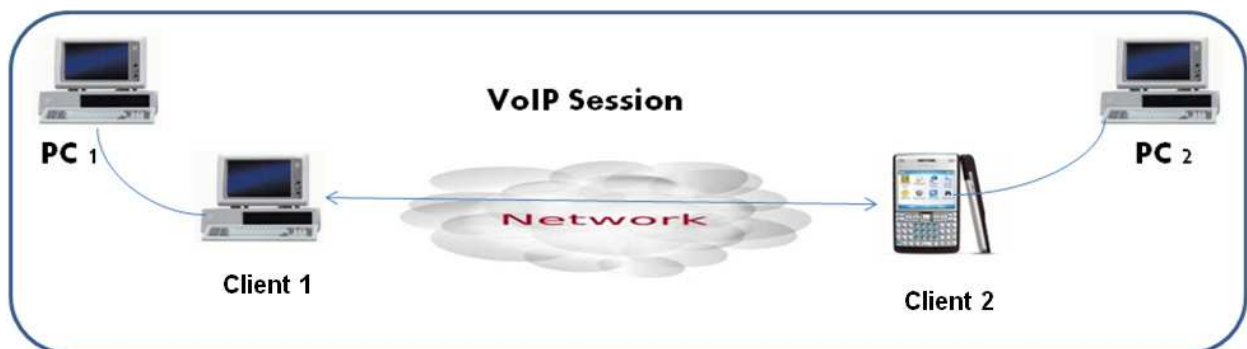


Fig. 7. Voice quality experiments setup [1]

It is reported that the voice quality is affected mainly by four different parameters [1]: The audio volume of reproducing device, CPU load, end-to-end delay, packet loss, as well as audio cables. For the first parameter, different tests provide different PSEQ scores through mainly calibrating the volume level. Hence, it is of interest to repeat the tests focusing on volume calibration for maximizing the PESQ scores. Considering the second parameter, it is proven that the CPU load is the major factor that degrades the performance of perceived voice when Skype client is a mobile terminal and this factor is dependent on the applications running on the mobile terminal. During experimentation, the Pocket Hack Master application is used to check the CPU load of SPV M5000 PDA. For end-to-end-delay, which is not involved in PESQ evaluation, the measurements that are taken into consideration are those which have end-to-end delays below 150 ms where the interactivity is guaranteed

leading to complete concentration on the audio signal quality. For the last parameter, it is proven through a set of experiments that different PSEQ results are obtained when audio cables get varied and this is due to the following: material, process of digital-to-analog conversion, analog audio transmission, and analog-to-digital conversion.

It is interesting to discuss some of the obtained results of the conducted experiments for evaluating the perceived voice quality of the Skype calls. Table 1 shows the MOS scores when considering the scenario where the Skype client is a PC-based node. The MOS scores are presented for different network environments, namely, ADSL, LAN, and WLAN. It is observed from this table that the achieved MOS scores in all networks are above 3.6 which reflect acceptable perceived voice quality. It is also shown that the differences between MOS scores in all networks are small and this is expected since the throughput of a Skype call is much less than the available bandwidth in all networks' connections.

| Network Environment | Uplink Traffic | Downlink Traffic |
|---------------------|----------------|------------------|
| ADSL | 3.989 | 3.935 |
| LAN | 3.838 | 3.838 |
| WLAN | 3.631 | 3.631 |

Table 1. MOS scores when running Skype client as PC-based nodes

Table 2 is done to provide MOS scores for different networks' interfaces WLAN and Universal Mobile Telecommunications System (UMTS); one of the mostly widely used 3G wireless technology in Europe, when a Skype client is a mobile terminal (SPV M500 PDA). As mentioned earlier that the CPU load has a great impact on the quality of Skype calls. Therefore, Table 2 provides MOS scores for CPU load ranges from 66% to 81% in WLAN and UMTS connections, respectively. Results show the all obtained MOS scores are not promising since they fall below 3.6. Hence, the voice quality is poor (unacceptable) compared with the results discussed in Table 1. It is interesting to see that during the traffic from PDA to PC, the MOS score is about 3.39 while it is, in the reverse traffic, about 2.85. Actually, this is due to the effect of many factors, including terminal hardware, operating system, and the running Skype version. One more observation can be extracted that the MOS score in the reverse traffic of UMTS interface is higher than that of direct traffic and this is due to the way the operating system interact with UMTS cards.

Also it is important to mention that according to the results of Tables 1 and 2, one can notice that the voice quality for the wired network utilizing ADSL and LAN networks was better than the wireless counterparts (WLAN and UMTS). Furthermore, the voice quality for the WLAN was much better than the UMTS case. One reason for that since calls over this network will traverse larger distance in the wireless channel which makes it more prone to interferences and errors, while the signal will traverse smaller distance in the wireless channel when the WLAN is used.

| Network Environment | Uplink Traffic | Downlink Traffic |
|---------------------|----------------|------------------|
| WLAN | 3.390 | 2.850 |
| UMTS | 2.636 | 2.873 |

Table 2. MOS scores when running Skype client as a SPV M5000 PDA

Table 3 presents MOS scores for the same networks' interfaces used in Table 2 but with more increase in the CPU load of both connections, as a range from 83% to 90%. When comparing this table with Table 2 and considering WLAN, The MOS scores are decreased to 2.496 and 2.477 for uplink and down link traffic, respectively. These scores are also decreased, when using UMTS connection, to 2.617 and 2.66 for uplink and downlink traffic, respectively. This is due to not only the hardware resources, but also the terminal processing which lead to getting poor perceived voice quality. Furthermore, it is noticed that the MOS scores for UMTS connection exceed those obtained for WLAN connection. This can be justified through the better optimization made in processing of SPV M5000 terminal when using UMTS.

| Network Environment | Uplink Traffic | Downlink Traffic |
|---------------------|----------------|------------------|
| WLAN | 2.496 | 2.477 |
| UMTS | 2.617 | 2.660 |

Table 3. MOS scores when running Skype client as a SPV M5000 PDA (with increase in CPU load)

3.2.3 Concluding remarks and recommendations

As a summary, a comprehensive study is made to evaluate the Skype calls, using PSEQ method, over fixed and mobile network connections, namely, LAN, WLAN, ADSL, and UMTS. Many factors may affect severely the performance of Skype calls if it is not investigated and designed properly, including audio volume, CPU load when PDA used as a mobile terminal, end-to-end delay, as well as audio cables. Another important factor comes from the fact that the current Skype application used for the wireless environment does not take into account the additional channel impairments such as bit errors due to wireless channel interference which in turn will increase the packet loss. Indeed, this additional challenge did not exist in the wired environment, so it should be also taken into account when designing a high-quality VoIP application. Results show when using a Skype client as a PC-based node that the quality of perceived voice is much better than the one obtained when using that client as a mobile terminal. Therefore, current VoIP applications that run on fixed networks need to be enhanced and optimized more so that their performance over wireless networks is not degraded.

3.3 VoIP over wireless challenges

In what follow, we describe the main challenges associated with wireless transmission and how they affect the transmission quality. Namely, we will focus on bit errors introduced by other interfering signals, packet loss due to buffers overflow of the intermediate nodes, and finally, we describe the importance of having efficient and effective CAC algorithms and how the current algorithms can be further enhanced to improve the network utilization and voice quality.

3.3.1 Bit errors due to interference

A Radio Frequency (RF) transmitted over the open space maybe interfered by any other RF signals within the same frequency range, which may cause bit errors to the original signal,

which in turn corrupts the transmitted packets. Furthermore, it is important to demonstrate what do we mean by bit errors, which means simply that some bits flip from zero to one or vice versa. In other words, the receiver still receives the entire transmitted sequence of bits but not exactly like the same transmitted status. It is also important to emphasize that bit errors do mainly exist in the wireless media and not in the wired one due to the natural of the wireless media which makes the transmitted signal more prone to any other interfering signals, while in the wired media, the signal is protected from these interfering signals by the wired media. In case of wireless transmission, interference may occur from any other electronic devices that may emit noise such as microwave ovens, cordless phones, and the legacy wireless LAN devices.

Interfering signals are not permanent and may exist in certain periods of times. This can be modeled as a two-state Markov Chain model that is shown in figure 8. In this model, the wireless channel is represented into two states; good and bad. In good channel conditions, there is a big probability to receive the bits without any errors, while the bits are more likely to be received in error while being in the bad state. Moreover, the channel conditions alternate between the good and bad states with probabilities P_1 , P_2 , respectively, and will remain in the good or bad states with probabilities $1-P_1$, $1-P_2$, respectively.

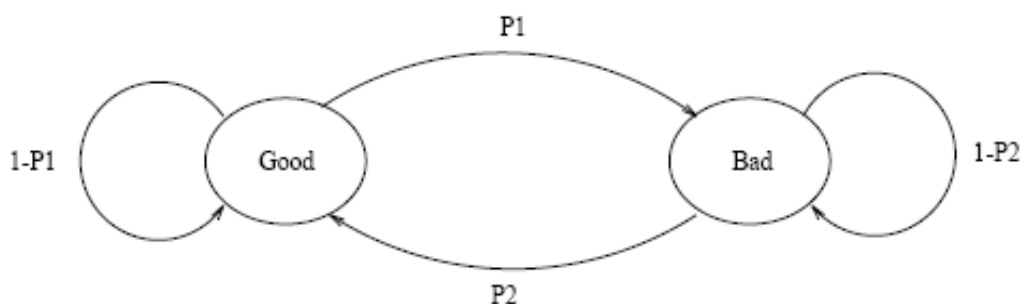


Fig. 8. A two state Markov model for the wireless channel

3.3.2 Packet loss due to congestion

Transmitting voice packets over wireless eventually will involve communicating with the wired network. As shown in figure 4, this backhaul network may introduce packet losses due to the buffers' congestion of the intermediate routers and nodes. It is also important here to distinguish between the natural of bit errors introduced by the wireless network and the packet loss introduced by the wired network. As described earlier, bit errors will simply flip some of the bits, so the stream of bits is completely received but not necessary in the same sent structure. For example, if the sent bit stream was 1010101010, the received stream maybe like this: 1010101111. Notice that some bits (bit number 8 and 10 in this example) flipped from 0 to 1. While in packet loss, the entire packet is lost and dropped due to the fact that the intermediate routers' buffers may not have enough space to accommodate any new arriving packets. Consequently, at the receiving end, we may not have the same number of the transmitted packets.

3.3.3 Call admission control

One of the most important factors that affect voice call quality is the number of calls admitted to the wireless network. Wireless network has certain bandwidth and capacity and

if the number of admitted voice calls exceeds its capacity, calls start to degrade in quality and users will not be satisfied. As such, CAC algorithms are often introduced to limit the number of admitted voice calls according to the wireless network capacity and bandwidth. Furthermore, CAC algorithms are also implemented in the wired backhaul network, more precisely in the VoIP gateway, which is responsible to establish an SIP connection to the destination node. CAC in this stage works by probing the VoIP call path, i.e. sending probe packets to the destination VoIP gateway, and measures the parameters that affect the transmission quality such as the packet loss, end-to-end delay, delay variation (jitter), and other network parameters that directly affect voice quality which will be used in the admission process. The author of [12] explored the fact that in wireless WiFi VoIP, there are two separate CAC algorithms running; one in the wireless network, which is responsible for admitting voice calls over the wireless domain, and the other one in the wired network that admit/reject the call based on the QoS status of the network path that the voice call is expected to follow. As we can notice, both CAC algorithms run independently and without any coordination with each other which may lead to a low network utilizing. This may occur due to fact that calls maybe accepted by the wireless domain but eventually maybe rejected by the VoIP gateway due to the fact that the network paths of these calls maybe congested. As such, the author of [12] showed that better network utilization and better VoIP performance can be achieved if both CAC algorithms coordinate with each other, which will be positively reflected on the voice quality of the admitted calls. The proposed solution will be further described in the next section.

4. Opportunistic solutions

Most of the VoIP proposed systems and applications do not take into account the new challenges of the wireless network, so they just customize the interfaces and the design of their applications to make it work with different wireless mobile platforms and devices, or make minor changes in the implemented algorithms and structures. However, as depicted in the previous section, simple design and platform upgrades of the applications specifically designed to computers, laptops, mobiles, and handheld applications may not be sufficiently efficient to meet the expectations of the mobile phone users who expect a comparable VoIP quality to the typical mobile phone voice quality. Otherwise, these applications will not be appealing and will not constitute a valid and strong alternative to their typical mobile phone voice service. In this section, we describe the state-of-the-art solutions and algorithms proposed to compact bit errors and packet loss introduced by the wireless channel that can be integrated into the current VoIP mobile applications, and also how CAC algorithms can be further improved to achieve higher network utilization and better voice quality.

4.1 Solutions to bit errors and packet loss

When VoIP developers and companies started to provide VoIP services and solutions over the wired network, they were mainly concerned about protecting voice packets against packet loss introduced by the transmission media. As such, several mechanisms and methods were introduced to recover and combat these lost packets. Figure 9 shows two widely used mechanisms in many wired VoIP applications. We call the first scheme as the Packet-Level Forward Error Correction scheme (PL-FEC), where the second scheme is named as the Media-Dependent Error Correction scheme (MD-EC). Moreover, we are

assuming that in both packetization schemes, each packet contains only one voice frame. However, this assumption is accepted despite its high packetization overhead especially for real-time applications such as VoIP, in which end-to-end delay is very essential and need to be greatly minimized. Thus, putting one voice frame on each packet will reduce the end-to-end delay and the play-out time. In what follows, we will provide a brief description of each scheme and then we will discuss new proposed schemes and methods especially designed for the wireless network, which in turn provide better transmission quality and performance than the legacy wired applications.

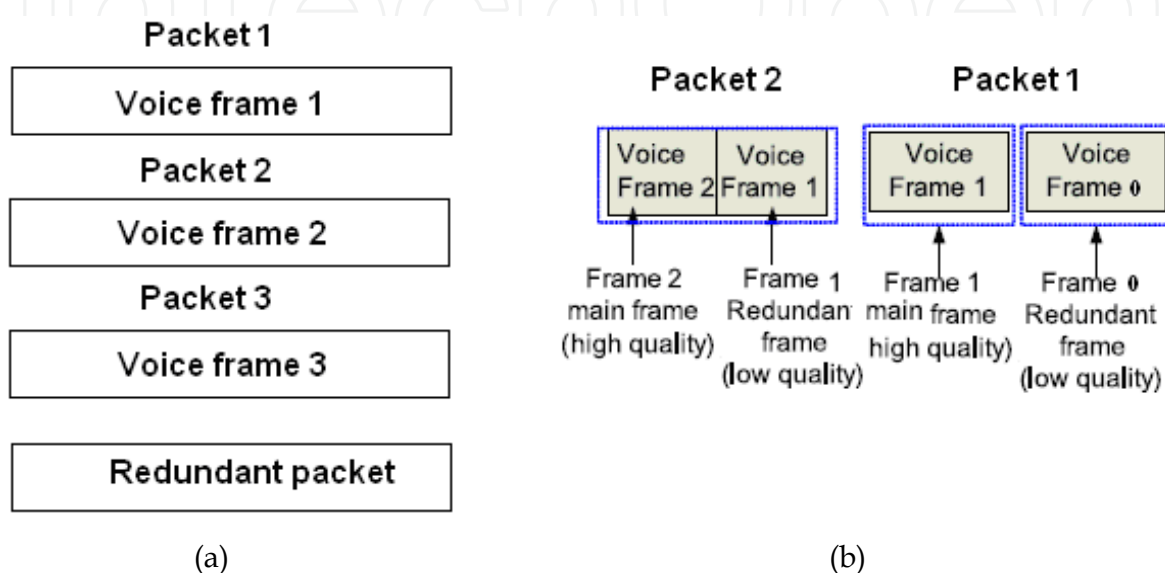


Fig. 9. Shows two error correction schemes used to protect against packet loss. (a) The Packet-level Forward Error Correction scheme, and (b) The Media-Dependent Error Correction scheme

4.1.1 Packet-level error correction scheme

The basic idea of this scheme is to group a small number of packets and do some operations on them to generate a redundant packet, then if any packet out of this group of packets is lost, then this packet can be recovered using the other received packets. Figure 9 (a) shows an example of this scheme, here a group of 3 packets are grouped together (packets 1,2,3), a redundant packet is composed by performing a bit-wise XOR-ing operation with the other packets, as such, losing any packet from the four packets can be easily compensated at the receiver side by XOR-ing the received packets. The main advantage of this scheme that it is simple and easy to implement. However, it is not very efficient, as the transmission overhead is high (in this case, the transmission overhead is 25%), and it works if only one packet out of the n -transmitted packets is lost. Also, the receiver has to wait till it receives the n -transmitted packets, which may increase the end-to-end delay.

4.1.2 Media-dependent error correction scheme

The second mechanism widely used to mitigate packet loss is depicted in Fig 9 (b). In this scheme, each voice frame is encoded into high and low quality encoding rates. An example of a high quality encoded rate will be to encode the voice frame using 13 Kbps encoding

rate, while the encoding rate of 5.3 Kbps can be used for the low quality encoded voice frames. Once the voice frames are encoded, the high quality encoded frame is packetized in one packet, and the lower quality version is packetized in the packet immediately after it. Now, if the packet containing the high quality frame is lost, and the packet containing the low quality is received, then the receiver uses the low quality packet to conceal the lost high quality voice packet. If both the low and high quality packets are received, the receiver uses the high quality packet, if both packets are lost, the receiver uses other received packets (normally the one after or before it) to conceal the lost packet. For example, Figure 9(b) shows an example of this packetization scheme, packet 1 contains two voice frames, one high quality voice frame (voice frame 1), and the a low quality frame (voice frame 0), the low quality version of voice frame 1 is packetized in a different packet (packet 2), so if packet 1 is lost and packet 2 is received, then the low quality version of voice frame 1 will be used to substitute the lost high quality voice frame.

4.1.3 PL-FEC and MD-EC for wireless VoIP

After discussing the two most widely used error correction schemes for wired VoIP applications, we now discuss the deficiencies of these mechanisms in case of having bit errors introduced by the wireless channel. The main weakness point here that in case of bit errors, the entire packet will be corrupted and dropped even if one bit was in error, even worse, if bit errors were distributed among several packets, all packets will be redeemed lost and none of the above packet level error correction schemes will work. The only solution to mitigate bit errors is to use error correcting codes that can be applied on the bit-stream. These error correction schemes work by dividing the bit stream into a group of symbols of a predetermined size, where the symbol of size (s) is simply a group of s bits, then certain polynomial operation can be applied to generate some redundant symbols that can be used in detecting and correcting random bit errors. One widely used error correction code is Reed Solomon codes [14]. In this book chapter, we will describe how using these techniques can greatly improve the performance of wireless VoIP transmission. However, describing how error correction and detection codes work is beyond the book chapter scope and we encourage the reader refer to other resources such as [13,14] for further information.

4.1.4 A hybrid media error correction scheme for wireless VoIP

In this subsection, we describe our novel approach in solving the problem of transmitting voice packets over wireless channel susceptible to both types of errors; bit errors and packet loss. Knowing that we greatly encourage the reader to refer to [15] for more details and description.

Figure 10 shows our proposed approach for protecting packets against both types of errors. As shown in the figure, the proposed scheme utilizes the Media-Dependent Error Correction scheme described in the previous subsection for providing protection against packet loss. For example, each voice frame is encoded using two different encoding schemes, one for high quality and another one for low quality version of the audio frame. The low quality encoded version is considered as a redundant copy of the high quality frame, and it is used to substitute the high quality frame if it is lost. Now, in order to protect packets against bit errors, we have proposed to use error correction and detection codes such as the Reed-Solomon codes on the packet level, so each packet is divided into a group of symbols, and a

polynomial function is used to generate redundant symbols that can be set by the algorithm, which controls the error correcting capabilities of this packet such that the higher the redundant symbols, the larger number of errors can be corrected using this error correction scheme. In figure 10, these redundant symbols are described using the symbol RS CC, which stands for Reed Solomon Channel Code. However, the higher the redundant symbols, the higher the transmission overhead, so it is at the end a trade-off between the error correcting capabilities and the introduced overhead. So in this scheme, if bit errors happened, then using the Reed-Solomon codes, these errors can be corrected instead of just dropping the entire packet. In our proposed approach [15], we even take into account applying unequal error correction symbols to each packet according to the perceptual importance of each packet. In other words, voice packets are different on their perceptual importance, some of them may contain higher and larger energy than other silent or unvoiced packets, as such, losing these packets may greatly affect the quality of the entire voice call, while losing less important voice packets may have lower impact on perceiving voice calls. Moreover, the location of the voice frame determines its perceptual importance, for example, voice frames that are located in the transition portions from voiced to unvoiced and vice versa are more important than non-transition frames. As such, more protection (means higher number of redundant symbols) should be applied to these important packets. We highly recommend the reader to refer to our work for more details on how we determine which packet is more important and how we distribute the redundant symbols among different packets such that the expected received quality of the group of packets is maximized.

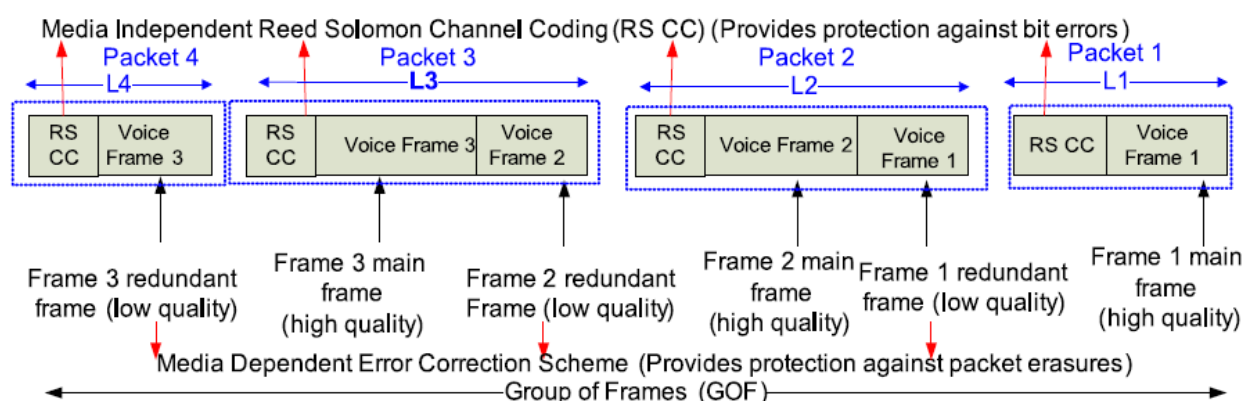


Fig. 10. Hybrid Media Error Correction scheme [15]

Furthermore, we would like to mention why this scheme is named as a hybrid scheme. This is due to the fact that it uses two different types of error corrections schemes, the one used for detecting bit errors, which is considered as a media-independent error correction scheme, while the other one which is used for correcting packet loss which uses a media-dependent low quality copies of the original voice frames. It is also worth mentioning that the proposed scheme utilizes the power of using Multiple Input Multiple Output (MIMO) antennas which has the effect of further reducing bit errors introduced by the wireless channel, and gives more protection to the transmitted bit stream.

Finally, figure 11 shows a performance evaluation results of the proposed Hybrid Media-Error Correction scheme (HM-EC) and compares it to other schemes that are designed mainly for the wired transmission media and can only provide error correction capabilities

against packet loss. The experiment setup was as follows. First, the speech signal is divided into a group of voice frames (GOF), each GOF is analyzed and processed separately. Then, each voice frame is encoded using the Speex encoder [16] into low and high bit rates, then the high encoded frames are packetized into one packet and the Reed Solomon error correction scheme is applied to each packet. The number of redundant symbols used for each voice frame is determined based on solving an optimization problem fully described in the paper, which takes into account multiple parameters such as the wireless channel quality, the packet loss, and the perceptual importance of each voice frame. Then, the low quality frames are packetized with the subsequent packet along with the higher quality version of the next frame, and Reed Solomon Channel Coding symbols are applied to the entire packet. To simulate bit errors and packet loss introduced by the wireless channel, two different independent models for the wireless channel were used, the first one is the Gilbert-Eliot model used to simulate bit errors, while the second one is the Gilbert model used to simulate packet loss. More details about the specifications and the mathematical details of the wireless channel model are available at [17].

Once bit and packet errors are simulated at the transmitted stream, the received stream is decoded using the Speex decoder, then error concealment is applied to conceal the lost and non-correctable packets, and then the received voice packets are played out. However, in order to evaluate the quality of the received voice sequence, the received packets were saved into a WAV file and compared with the reference transmitted voice signal using two different metrics, the first one is the Log Spectral Amplitude Distortion (LSAD), defined in [15], where lower LSAD values mean better voice quality, and the second metric is the Perceptual Evaluation of Speech Quality algorithm [12], where higher score indicates better voice quality. In the performance evaluation figures below, one can notice that the x-axis represents different values of the wireless Signal to Noise Ratio in the Good state (SNR_G) measured in dB. Lower values indicate bad wireless conditions channel and higher bit errors, while higher SNR_G values indicate better wireless channel conditions and less bit errors. In the conducted experiments, we varied the packet loss ratios and the bit errors into different rates, and in order to verify the effectiveness of our approach, we compare the performance of our approach with two other approaches. The first one is called the Optimal Unequal Protection scheme (OUP), and the other one is called Optimal Piggy-backing Error Correction scheme (OPEC). The OUP scheme relies on protecting the most perceptual voice packets by utilizing the packet level error correction scheme described earlier. This scheme is a modified version of the work proposed by [18]. In this scheme, the expected distortion for losing each voice packet is calculated, and the ones with highest distortion among the other packets in the Group of Frame (GOF) are protected using the packet level forward error correcting scheme. The number of redundant packets per GOF depends on the allocated transmission budget. While the OPEC utilizes the adaptive piggy-backing error protection policy used in many VoIP applications such as the Robust Audio Tool (RAT) [19]. In this scheme, the sender protects the most important voice packets by providing a low quality version of these voice packets piggy-backed with the following packet, so if the original voice packet get lost and the low quality packet was received, then the receiver uses the low quality version to conceal the lost packet. The sender determines the most important packets by calculating the expected distortion of losing each packet within the group of frames, and then depending on the transmission budget, it chooses the packets with the highest distortion and marks them as the most important perceptual packets.

As can be seen from figure 11, the performance of HM-EC which is aware of the wireless channel impairments and can provide protection to both types of channel errors; bit errors (utilizing the Reed Solomon error correction scheme), and packet loss utilizing the MD-EC scheme, has lower distortion and higher PESQ values for different SNR_G values, and outperforms its counterparts (OUP, and OPEC) who can provide protection to packet loss only and not well suited for the wireless transmission media.

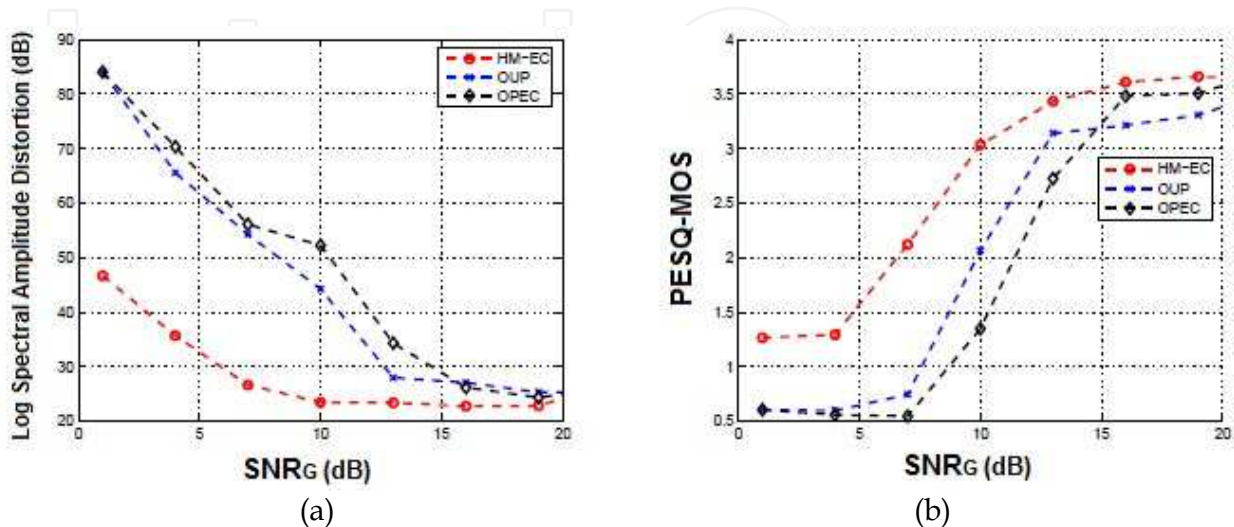


Fig. 11. Performance evaluation results of (a) LSAD as a function of SNR_G , and (b) PESQ-MOS as a function of SNR_G . The dg145 speech clip [23] is used to simulate a VoIP call with a transmission budget of $BT = 8,056$ Kbytes utilizing 2×2 MIMO link, the average packet loss rate is set to 10% [15]

4.1.5 Optimal rate adaptation for VoIP over wireless

Another approach proposed in the literature that provides efficient protection for voice packets against bit error and packet loss is based on utilizing variable rate voice encoders that can encode the voice frames using different encoding rates. The main idea here is to divide the transmission budget (the amount of bytes the sender is willing to spend to transmit the VoIP call), into two portions, the source code portion; the bytes used to encoded voice packets, and the channel coding budget; the bytes used to provide error protection such as the Reed-Solomon error correction codes. The main challenge here is to determine the best splitting ratio of the budget between the source and channel budgets. This problem is known in the literature as the joint source-channel coding, and it is an optimization problem in which increasing the source coding budget in one hand, improves the voice quality and reduces the coding distortion. On the other hand, the channel coding budget decreases, which in turn reduces the error correcting capabilities. So it is a trade-off between voice quality and error correcting capabilities. This problem was tackled and a proposed solution was suggested in [20]. In what follows, we briefly describe the proposed solution but we highly encourage the reader to refer to the paper for more details.

Figure 12 shows how this scheme works. First, voice frames are divided into group of frames (M) named as a Talk Spurt (TS), these voice frames are analyzed and processed to determine what is the optimal encoding rate (R_1, R_2, \dots, R_O) for each voice frame (F_1, F_2, \dots, F_M). The decision to choose which encoding rate should be assigned to each voice frame

takes into consideration many parameters and issues such as: the perceptual importance of each voice frame, the wireless channel conditions, and the network packet loss rates. All of these factors are quantified and inserted into a mathematical objective function that aims at maximizing the expected quality of the received voice packets. The optimization problem is solved and the optimal encoding rates of each voice frames are determined. Notice that once the encoding rates are determined, the amount of channel coding symbols that can be allocated to each frame is also determined. This fact becomes clearer if we look at figure 12 (b), which depicts the packetization process. First, the symbols S_{xy} , and C_{zh} refer to the source coding and channel coding symbols, respectively. Each column represents a channel coding block of size (L), so as we can see, the number of source coding symbols (S_{xy}) and channel coding symbols (C_{zh}) for each voice frame should be equal to L. For example, the first column represents the source and channel coding symbols that belongs to the first voice frame, so the subscripts x, and z in the variables S_{xy} , and C_{zh} refer to the frame number, where the subscripts y, and h refer to the number of source and channel symbols allocated to each frame, respectively. For example, in this figure frame 1 was allocated three source coding symbols (S_{11}, S_{12}, S_{13}), and three channel coding symbols (C_{11}, C_{12}, C_{13}).

As we can notice, the higher the encoding rate of each voice frame, the larger the number of source coding symbols S_{xy} , which results in a better voice quality and lower encoding distortion. However, the number of channel coding symbols C_{zh} will be lower for that frame, since the sum of S_{xy} and C_{zh} is equal to L, which in turn, will reduce the error correcting capabilities of that frame. So it is a trade-off between the voice encoding quality and the error correcting capabilities of each frame.

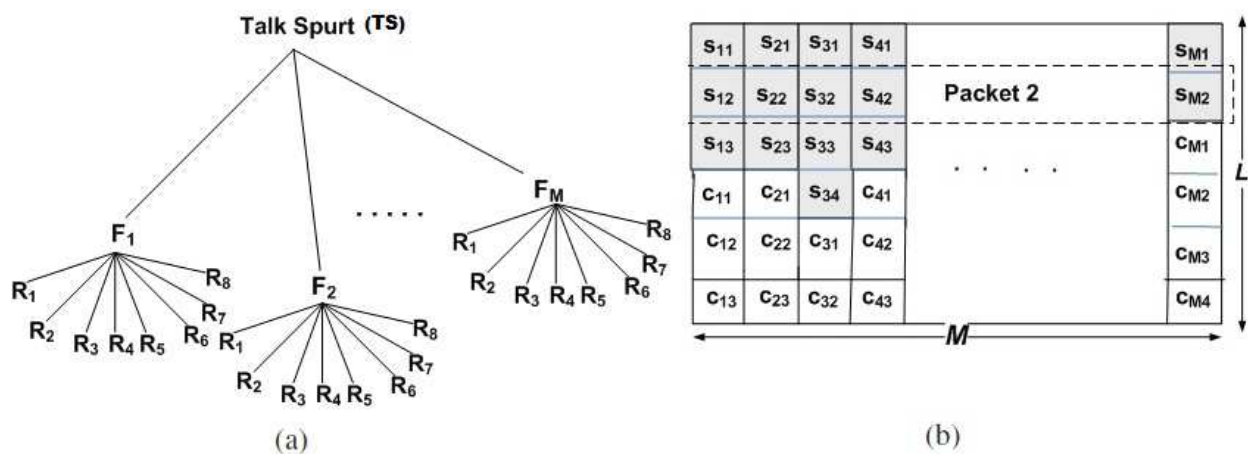


Fig. 12. (a) Shows how the frames that belong to a certain Talk Spurt (TS) are encoded differently using on of the possible encoding rates (R_1, R_2, \dots, R_8) supported by the Speex encoder, (b) shows how each voice frame is aligned vertically, such that packets are formed horizontally such that each packet contains one symbol from each voice frame [20]

Now once the optimal source and channel encoding rates are determined for each voice frame, the symbols are aligned into a two-dimensional grid of symbols as shown on figure 12 (b), where the encoded voice frames with their channel coding symbols are aligned vertically, and packets are formulated horizontally, i.e. the packet is formulated by taking one symbol from each voice frame. Notice that we suggested using the Reed-Solomon

channel coding scheme, which is capable of detecting and correcting symbol errors and erasures.

It is important here to mention how this scheme is capable of protecting packets against both types of errors (bits error and packet erasure). First, one can notice that the way the packets are formulated will help in mitigating packet loss, as can be seen, each packet is formulated by taking one symbol from each voice frame (from each column), as such, losing one packet will result in losing one symbol from each voice frame, which can be corrected using the channel coding symbols allocated to each voice frame. Also, the same applies for bit errors, these errors can be corrected utilizing the channel coding symbols as well.

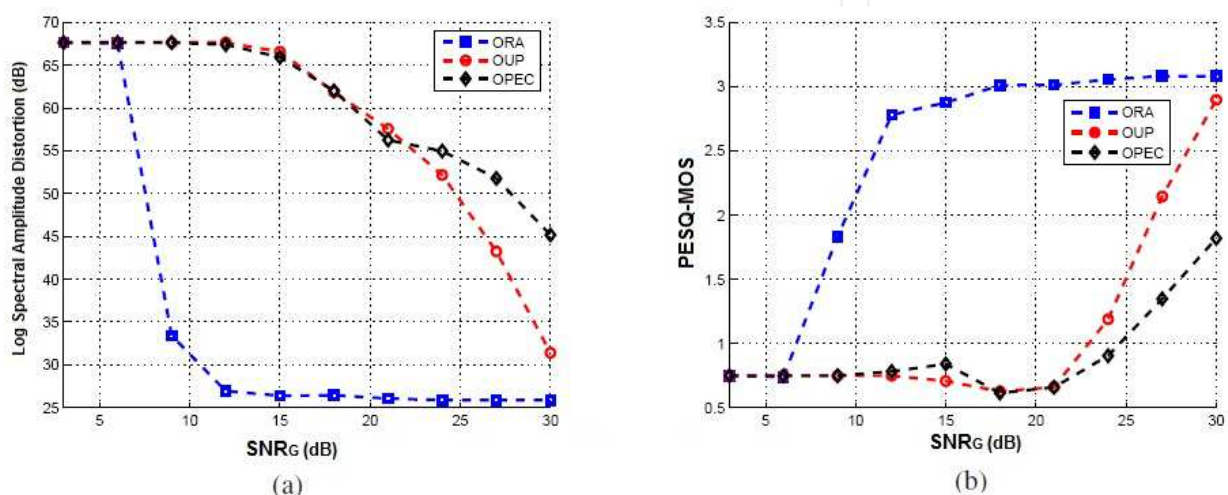


Fig. 13. Performance evaluation of ORA algorithm, using the dg129 speech clip [23] to simulate a VoIP call, utilizing 1x1 MIMO link configuration, with a transmission budget of BT = 5,6 Kbytes, and a 10% average packet erasure rate[20]

We conclude this subsection by showing the performance evaluation results of the optimal rate adaptation scheme under tandem wireless channel that is prone to bit errors and packet erasure. The same experiment setup used in the previous subsection is used here. Notice that again that ORA scheme has much better performance than the OUP and the OPEC schemes, especially at low SNR_G values in which the wireless channel is in bad condition. Once more, this scheme shows the importance of having error correction and protection scheme especially designed for the wireless channel, and also shows that other schemes especially designed for the wired network are not necessary efficient under noisy wireless channels.

4.2 End-to-end aware CAC algorithm

In this subsection, we will describe our proposed scheme published in [12], and show how it can improve the network performance and utilization, which in turn results in a better VoIP quality service and an increase in the number of successful VoIP calls conducted within specific period of time and within a specific network.

Probing the expected path of the VoIP call over the wired network is one technique widely used in many CAC algorithms at the VoIP gateways. In this technique, the end-to-end link status is used in determining whether the call can be admitted or not. However, as

mentioned earlier, in wireless VoIP service, we will have two stages for CAC, one at the wireless channel level, in which the wireless access point will check if the wireless link capacity can accept one more call without affecting the quality of the existing ones, and one at the wired level, that will check if the end-to-end path can meet the minimum VoIP QoS requirements. Currently, both algorithms do not interact with each other. We have proposed to exchange information between the two CAC algorithms and to use this information in the decision of the first CAC algorithm that takes place at the wireless domain. More particularly, we proposed to report the end-to-end link status information of each call discovered by the VoIP gateway to the Quality of Service Access Point too, so it can be used as a policy in the call admission process implemented in the wireless domain. According to this new setup, the QAP algorithm, will also check the link status of this call, reported by the VoIP gateway and use it to decide if the call should be admitted or not. If the information reported about the end-to-end path link of this call indicates that the destination path is congested and this call can not be admitted, the QAP rejects the call and does not register it in the Hybrid Coordination Controlled Channel Access (HCCA) polling list [21], consequently, the wireless resources will not be wasted in admitting unsuccessful VoIP call that will be ultimately rejected by the VoIP gateway. At the beginning, neither the VoIP gateway nor the QAP has entries in their link status table, so the QAP will admit at all calls, and the VoIP gateway will start building the link status table for these calls, so once this table is populated, it will be sent to the QAP to be used for the new incoming calls. The link information for the destination networks is stored in a lookup table that contains the destination addresses, and their paths statuses. So once the VoIP call is received by the QAP, its destination address is checked using that table, and the call is admitted only if its end-to-end path conditions are satisfactory. Caching concept is also used in our proposed scheme, so the link statuses information for the destination networks are kept in the cache memory for a certain time-out period, the VoIP gateway keeps updating the status of the destination networks attempted so far. After certain time-out period, the link status information related to a certain network destination is removed from the table, if no one accessed that network. Figure 14 depicts how our policy can be integrated with the QAP call admission algorithm.

To further show the benefits and the justifications of implementing this policy, let us look at a scenario in which the caller is trying to establish a VoIP call to a congested network. As discussed above, after the caller's terminal device establishes a connection with the access point, and registers itself as one of the admitted stations in the polling list, the caller will contact the VoIP gateway, and go through billing, authentication, authorization, destination IP address look up stages, and finally check whether the call can be admitted or not. In this case, the call is rejected and a busy signal is sent back to the caller terminal, which will try again to establish a connection and go through the previous process, until the destination link becomes less congested, so the call can be admitted, or until the caller gives up and stops from setting up the call. As we can notice that during the failed call setup attempts, the caller terminal utilized the wireless channel, and registers itself in the polling list for the whole period of time used in attempting to conduct the call. This time might be significant for the following reasons:

1. Contacting the VoIP gateway might take some time due to the fact that in real networks, it is usually hosted in the Internet Service Provider (ISP) network, not in the local wireless network.

2. Some time will be wasted for doing authentication and other functions mentioned earlier, which is considered useless as far as the call will be rejected, needless to say that these operations will consume some of the valuable VoIP gateway processing power and time.
3. The VoIP gateway might be loaded or its links might be congested, which might increase the process and access time and increase the delay in responding to the issued requests.
4. Usually, once the VoIP is rejected by the VoIP gateway, the caller station will still be registered in the QAP polling list, so the caller will keep trying many times to establish a call, and according to the network congestion statistics, the successive trials most likely will fail, since usually the congestion in networks lasts for a period of time [22]. While according to our policy, rejecting the call from the first time will give the opportunity to other stations to access the network, which might conduct VoIP calls with different destination networks that do not have congestion, thus improving the utilization of the wireless network, and increasing the number of successful VoIP calls.

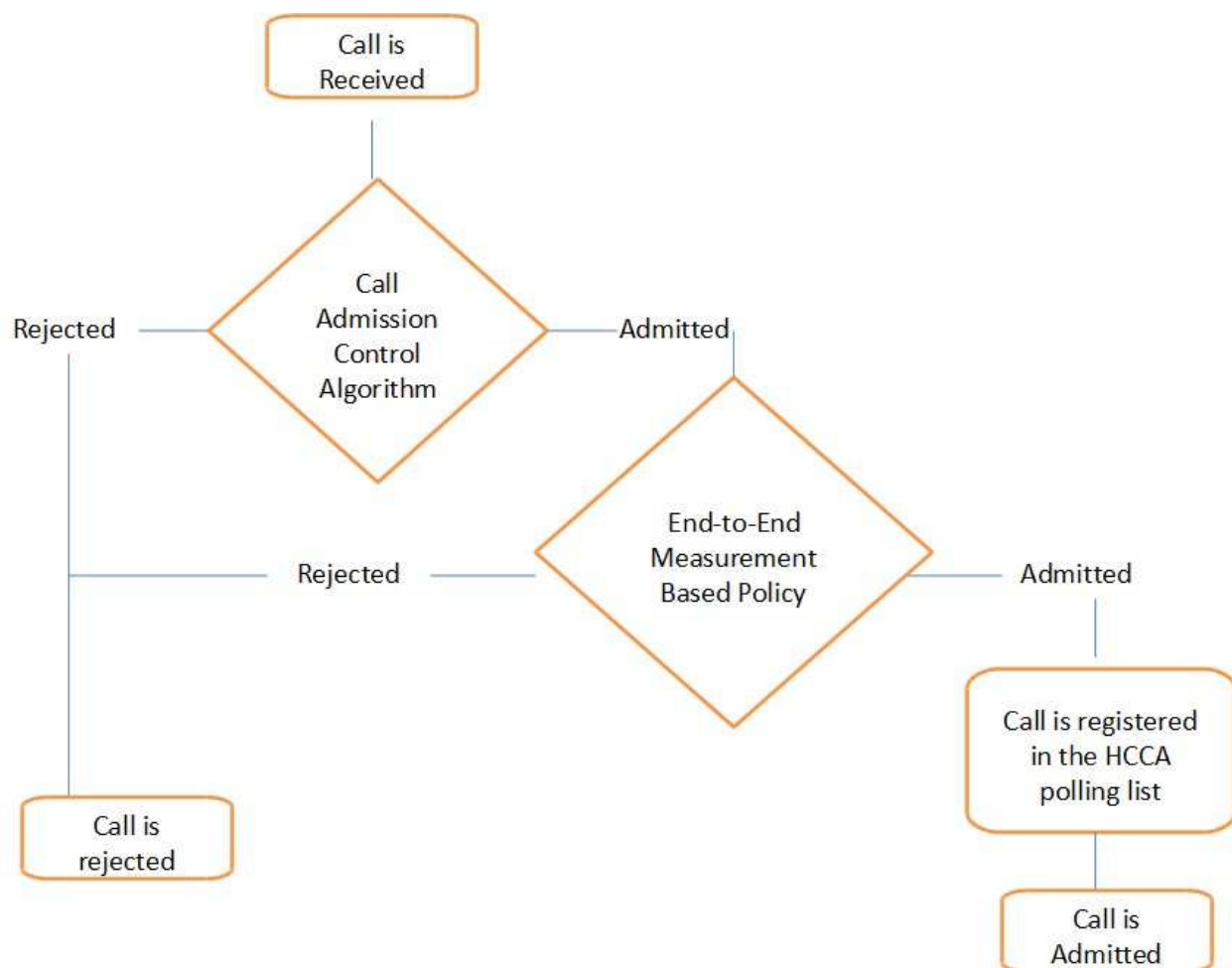


Fig. 14. Integrating the suggested policy with the typical QAP Call Admission algorithms [12]

5. Conclusion

VoIP over wireless is a promising and a challenging service that started to flourish between mobile and wireless users. Cost effectiveness and also the additional available services that VoIP can deliver are two driving forces that greatly motivate the Internet service providers and solution developers to develop new software and systems that are capable of delivering a competitive VoIP quality over the mobile network. Another great advantage that wireless VoIP can offer to its users is mobility. The mobile user will be able to use his own VoIP wireless number wherever he goes, in other words; the user will keep the same number even though the user may travel abroad or change the location. Different from the mobile phone, in this case, the user does not need to contact his mobile service provider to enable the expensive roaming service, all what is needed is to secure an Internet connection to the mobile phone. Also, recently, many mobile operators look at wireless VoIP service as a backup service for their customers when they do not have good mobile coverage at their homes for example or even in certain spots where there is not enough coverage from the mobile towers, so in this case, if the customer have an Internet connection, the user's mobile phone can connect to a device called femtocell [22], which will establish a connection with the mobile phone and conduct a VoIP call between the mobile phone and the mobile call center, where the call will be normally conducted as a normal mobile call to the final destination, on other words, wireless VoIP maybe used as an intermediate stage in many scenarios and solutions.

In conclusion, there are plethora of applications and opportunities for this service in the current and near future. As such, there is a need to develop new systems and applications to take into account the differences between the network impairments faced by the wired network, and the additional challenges faced by the wireless domain. In this chapter, we discussed some of these challenges, mainly bit errors introduced by the tandem wireless channel due to interference from other wireless devices, congestion and packet loss introduced by the intermediate nodes of the network, and the designing of an efficient Call Admission Control algorithm that takes into account the wired and wireless network status when admitting/rejecting VoIP calls. Another limitation briefly discussed in this book chapter is the relatively limited CPU power capabilities of the mobile devices and terminals, so the deployed algorithms and applications should take that into account. Furthermore, we have discussed some of the proposed solutions in the literature that can greatly improve the performance of wireless VoIP, namely we discussed two main approaches; the first one is called the Hybrid Media- Error-Correction scheme [15] and the second one is the Optimal Rate Adaptation scheme [20]. Finally, we discussed a proposed extension to the current Call Admission Control schemes used in the wireless domain. Our extension allow the wireless access point to interact with the VoIP gateway, which will eventually determine whether the VoIP call can be conducted or not, which is normally decided based on the expected end-to-end link status quality of each attempted VoIP call. We proposed to pass the link status information of the VoIP calls to the wireless access point and use it in admitting or rejecting VoIP calls over the wireless domain, which in turn will save a lot of bandwidth, improve the VoIP performance and increase the number of conducted VoIP calls in the wireless domain.

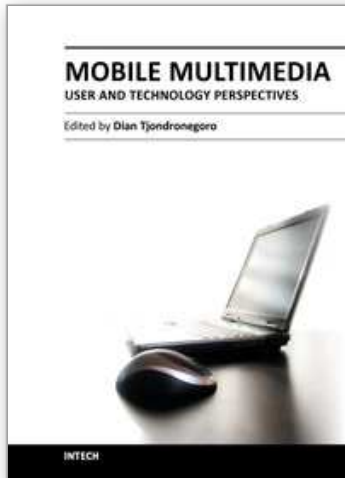
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As multimedia-enabled mobile devices such as smart phones and tablets are becoming the day-to-day computing device of choice for users of all ages, everyone expects that all mobile multimedia applications and services should be as smooth and as high-quality as the desktop experience. The grand challenge in delivering multimedia to mobile devices using the Internet is to ensure the quality of experience that meets the users' expectations, within reasonable costs, while supporting heterogeneous platforms and wireless network conditions. This book aims to provide a holistic overview of the current and future technologies used for delivering high-quality mobile multimedia applications, while focusing on user experience as the key requirement. The book opens with a section dealing with the challenges in mobile video delivery as one of the most bandwidth-intensive media that requires smooth streaming and a user-centric strategy to ensure quality of experience. The second section addresses this challenge by introducing some important concepts for future mobile multimedia coding and the network technologies to deliver quality services. The last section combines the user and technology perspectives by demonstrating how user experience can be measured using case studies on urban community interfaces and Internet telephones.

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