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White Paper

Including VoIP over WLAN in a Seamless Next-Generation Wireless Environment

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Overview

Seamless wireless data and voice communication is fast becoming a reality. In fact, the technology to enable one phone number for broadband wireless data and voice communication is available now. The remaining issues facing handset designers, carriers and service providers as well as enterprise and residential network designers relate to questions of deployment, configuration and network architecture. One key capability in the next-generation wireless world will be Voice over Internet Protocol (VoIP) using 802.11 wireless local area networks (WLANs). TI provides a comprehensive portfolio of wireless technology and the integration capabilities needed to support all aspects of a seamless wireless environment, which will certainly include VoIP over WLAN applications.

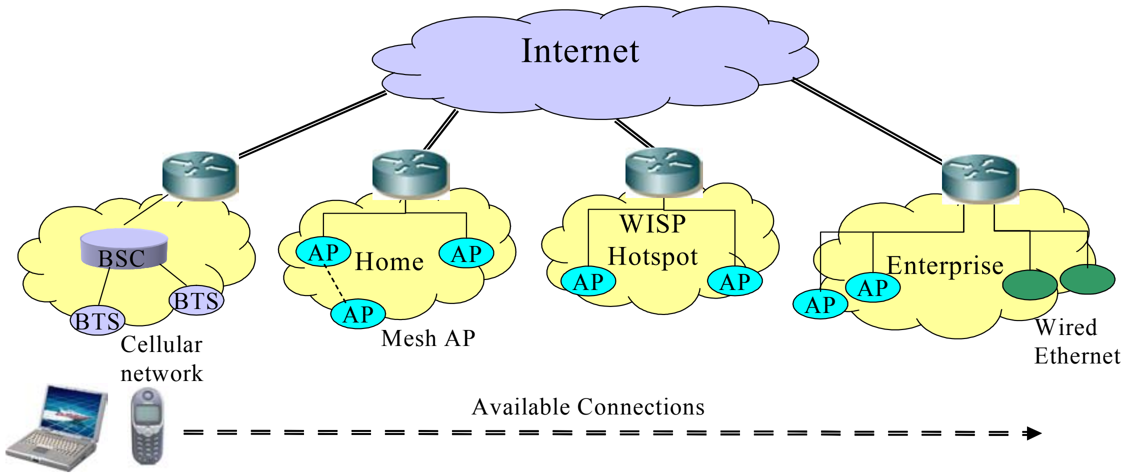
For wireless equipment manufacturers, service providers and enterprise/home network designers, VoIP over WLANs raises several deployment and planning issues concerning quality-of-service (QoS), call control, network capacity, provisioning, architecture and others. In addition, if the performance of each individual WLAN is to be optimized, these deployment issues must be addressed individually on a WLAN-by-WLAN basis. The requirements of the three main segments making up the WLAN marketplace also will have an effect on the deployment parameters of WLANs. These three market segments are:

- Residential/SOHO (small office, home office) cordless phones or scaled-down PBXs that will function as part of an integrated gateway
- Enterprise mobile VoIP WLAN network (private network)
- Cellular off-load network (VoIP over WLAN in hot spots, which in turn interfaces to the public telephone network)

TI's WLAN technology and support resources are capable of providing advanced solutions that address the entire market's critical requirements.

This white paper offers an overview of VoIP over WLAN applications and explains several critical deployment issues. Crucial to the success of VoIP over WLAN applications will be the ability of WLAN technology to support and provision QoS capabilities. Further, voice services inherently involve call control signaling that requires a high level of priority in order to meet the timing constraints of interfaces to external networks, such as the wireless cellular network or the public switched telephone network (PSTN).

While deployment of the infrastructure needed for VoIP over WLAN applications will take some time to be put in place, the following diagram illustrates the goal of an IP-integrated network. Such a network would allow seamless multiple access options for most of the more prevalent voice and data services.



WLAN Network Capacity Analysis

For network planners who are deploying a VoIP over WLAN application, one of the first issues to be addressed should be network capacity. To ensure the network is able to deliver the required QoS capabilities for a voice application, designers must anticipate and analyze how the WLAN will be used. Several questions, such as the following, must be answered:

- What types of QoS capabilities will be deployed?
- How much network capacity must be set aside for these QoS capabilities?
- What is the projected growth rate for QoS capabilities on the WLAN?

The questions above are network design considerations for a variety of contemporary applications, including VoIP, video and other services requiring QoS capabilities.

The purpose of this discussion is to explore the various facets of network capacity planning for the future deployment of WLANs. While the intent here is to analyze VoIP-enabled systems, network designers should also expect a significant amount of multimedia traffic over home/SOHO WLANs as well as video conferencing traffic over enterprise WLANs.

The remainder of this section describes the following:

- Over-subscription of voice networks (voice concentration)
- Throughput requirements for typical voice, video and media applications using IP packet technology
- WLAN network capacity for enterprise applications
    - RF frequency planning and reuse for large network deployments
- WLAN network capacity for home applications
    - Consideration of wireless repeaters (mesh) to extend home coverage

Over-subscription of Voice Services (Concentration)

It is important for designers of VoIP over WLAN applications to understand some of the basic concepts that have been applied for years in the PSTN. A basic understanding of over-subscription, for example, can assist network planners who are evaluating network capacity for enterprise VoIP over WLAN applications.

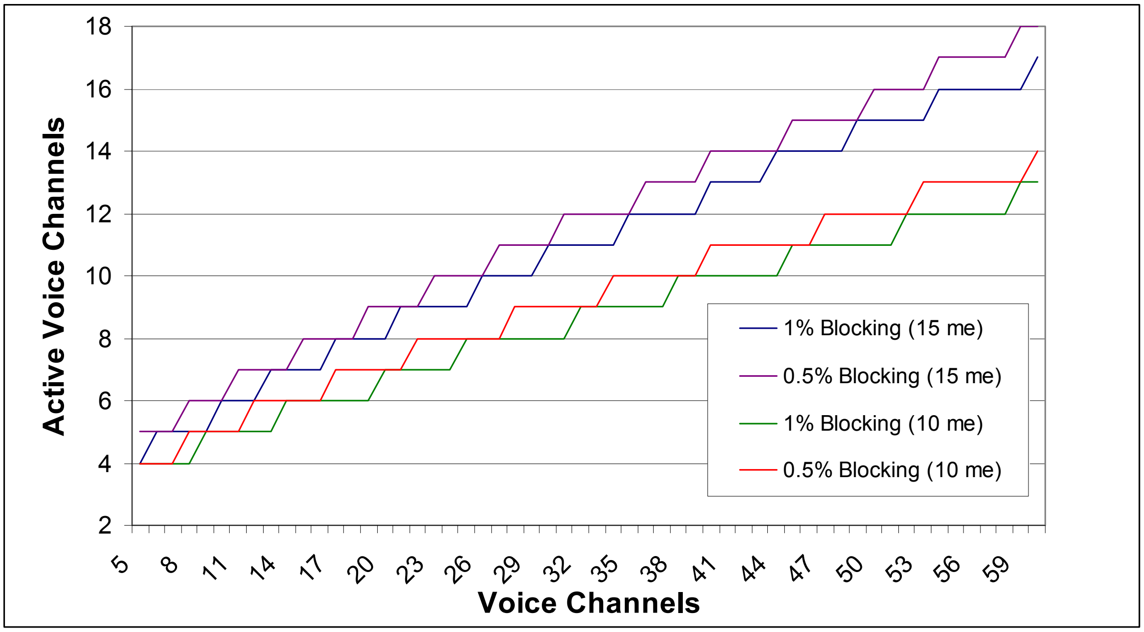
Telephone systems have been very closely monitored for over 100 years. The public telephone system has always incorporated “statistical over-subscription” of phone lines. In the United States, there are typically between four and eight phones per active (served) phone line in the network. POTS (plain old telephone system) networks are designed to have a specific probability that a call can be blocked from time to time. In the United States, call blocking is typically limited to 1% or 0.5% of total calls.

Phone lines typically are terminated at a Class 5 switch or a Digital Loop Carrier (DLC) connected to a Class 5 switch. The Class 5 switch manages call connections and rejects calls when the system capacity has been reached. (A caller is aware of this when receiving a fast busy tone or the “all circuits busy” message.) In cellular networks, some consideration is given to reserve a fraction of the active phone line capacity for handoff purposes between one cell and the next.

A measure of phone usage capacity is the ERLANG function, which equates to one active call hour (or 3,600 call seconds) of voice line use. The amount of phone concentration (over-subscription) can be determined with the ERLANG-B function, (the ERLANG blocked call function). Because the telephone network must be designed for the worst-case load, phone usage is defined as that level that is achieved during the busiest hour of the day. Accurate average measurements for peak busy hour phone usage in the United States are as follows:

- 0.15 ERLANG (15 me) for a business phone
- 0.1 ERLANG (10 me) for a residential phone

Based on the ERLANG B function and an acceptable percentage of blocked calls, the following diagram illustrates the number of active phone lines needed to support a set of phone users attached to a given switch or bandwidth resource.



As the pool of attached phone lines increases, efficiency, in terms of fewer blocked calls, and over-subscription also will increase. This should not be surprising because the efficiency of all systems that use statistical multiplexing improves as the number of channel resources increases at the multiplexer. The concentration level moves from around 2:1 at 10 subscriber phone lines to more than 3:1 at 60 user phone lines.

Voice, Video and Media Throughput over IP

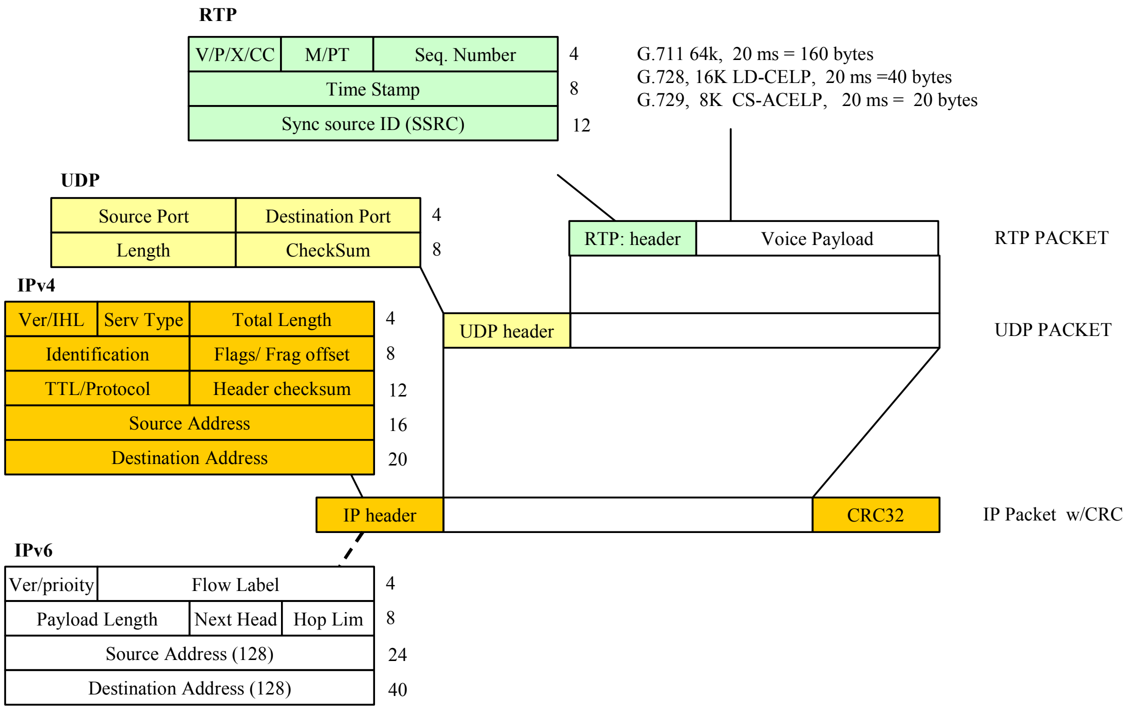
The following sections discuss several WLAN network capacity issues as they relate to the transmission of voice, video and multimedia data using IP.

Voice Compression and VoIP

Voice compression algorithms help network designers derive as much capacity from an infrastructure as possible, but compression algorithms involve tradeoffs between efficiency and overhead that planners should consider.

In wireless networks, voice is digitized with the G.711 coding standard and transported at 64 Kbps. While G.711 is the mainstream digital codec for toll-quality voice services, a number of more efficient codes are used for both cellular and voice "pair gain applications." In an IP network, voice codecs are placed into packets with durations of 5, 10 or 20 msec of sampled voice, and these samples are encapsulated in a VoIP packet.

The following figure illustrates the encapsulation for various protocols, including IPv4, UDP and RTP (Real Time Protocol). For IPv4, the packet overhead is 40 Bytes. As the industry transitions to IPv6, this overhead will grow to 60 bytes.



Clearly VoIP has an overhead issue that is compounded when high levels of voice compression are deployed in conjunction with voice packets of short duration. The tradeoff between overhead and packet duration is shown in the following table. Other issues affecting VoIP network capacity planning, such as delay, jitter and packet duration, are discussed later in this white paper.

	Voice Packet Frame Duration (msec) efficiency			
CODEC	5	10	20	40
IPv4 G.711	47.6%	64.5%	78.4%	87.9%
IPv6 G.711	38.5%	55.6%	71.4%	83.3%
IPv4 G.726	31.3%	47.6%	64.5%	78.4%
IPv6 G.726	23.8%	38.5%	55.6%	71.4%
IPv4 G.729	10.2%	18.5%	31.3%	47.6%
IPv6 G.729	7.2%	13.5%	23.8%	38.5%

The following table lists the one-way throughput requirements for typical voice codecs using VoIP. For the purposes of capacity analysis, a typical throughput of 64 Kbps per direction was used, assuming a combination of G.726 and G.711 codecs.

Coding algorithm		Bandwidth	Sample	Typical IP bandwidth (one way)
G.711	PCM	64 kbps	0.125 ms	80 kbps
G.723.1	ACELP	5.6 kbps	30 ms	16.27 kbps
		6.4 kbps		17.07 kbps
G.726	ADPCM	32 kbps	0.125 ms	48 kbps
G.728	LD-CELP	16 kbps	0.625 ms	32 kbps
G.729(A)	CS-ACELP	8 kbps	10 ms	24 kbps

<u>VoIP Complexity Options</u>

To deploy VoIP WLANs, two tiers of voice-capable access points (APs) probably will be needed:

- Low-end consumer VoIP APs will use G.711 and/or G.726.
- APs for the enterprise and wide-area applications will support a full suite of possible cellular and standard codecs for a wide variety of user devices such as PDAs and others.

To achieve completely seamless ubiquity of IP services, even low-end APs must support handoffs from cell phone traffic as well as a full set of codecs.

Video Media over IP

Although voice will be the first application requiring QoS capabilities over WLANs, several other multimedia applications will soon follow, including the distribution of audio (net radio, MP3 music, etc.) and video (streaming video, DVD, HDTV, etc.) over WLANs. Fortunately for network planners, media compression codecs will ease the bandwidth requirements for these multimedia applications. Specifically, improvements in the quality of video codecs like MPEG4 will allow DVD-quality compression at throughput rates of approximately one Mbps. For HDTV, the standard MPEG2 video stream can be reduced from 19.2 Mbps to around eight Mbps.

The following table illustrates the one-way throughput for various consumer video codec devices using maximum IP packet lengths.

Video Media	Bandwidth (Mbps)	Packet size	Packets/sec	Delivered Bandwidth (Mbps)	Overhead (%)
MPEG-1 Basic Media	2.5	1,500	2083	2.5663	2.58%
MPEG-2 SDTV format (DVD)	8	1,500	6666	8.2125	2.59%
HDTV MPEG-2 committee 18	19.2	1,500	16000	19.7120	2.60%

It should be noted that the FCC has mandated that by 2006 all televisions sold in the U.S. must include digital receivers. At that point, the integration of wireless interfaces into television electronics could be widespread.

Video Conferencing and IP Streaming Media

WLAN planners and designers should realize that video conferencing is an application that will have an impact on WLAN network capacity even though video conferencing has not yet become as pervasive in the enterprise or home markets as had been expected. This will change over the next few years as broadband connections become pervasive in households, the number of telecommuting workers increases, and enterprises improve their IT resources to allow greater use of video. As a result, WLAN designers should consider the requirements of video conferencing as they deploy infrastructure.

The following table provides a summary of the throughput requirements for several typical video conferencing and streaming media applications. Similarly to DVD and HDTV, the same types of improvements in lower resolution video and conferencing compression are expected in the years ahead. Network capacity models, especially in home networks, should anticipate increasing use of these applications.

Video Product	Packet Size	Bandwidth	Packets/Sec	Delivered Bandwidth	Overhead (%)
Business-Quality Conference	915	781,693	107	735,466	6.3%
NetMeeting Video LAN	779	478,312	77	445,156	7.4%
NetMeeting Video DSL	363	187,726	64	159,800	17.4%
NetMeeting Video 28K	288	10,497	5	8,529	23.1%
Real Audio Radio	681	165,118	30	152,025	8.6%
Media Player 80K Stream	687	81,171	15	74,882	8.4%
Media Player 20K Stream	476	27,600	7	24,469	12.8%
Real Video 28K Stream	384	25,173	8	21,633	16.4%

Throughput of WLAN Access Points (AP)

To optimize the network capacity of a WLAN with a voice or multimedia application, network planners must give special attention to the throughput of the APs which govern how quickly data of any sort can be placed on the network.

The following two basic functions affect the throughput of an AP:

1. Area and modulation density supported by the cell
    a. Small cells can support high data rate modulations (peak rates)
    b. Larger cells will use lower rate 802.11 modulations and are an aggregate sum of areas covered and the modulation rate
2. The WLAN MAC protocols have the following effects:
    a. The Ethernet (CSMA/CA) protocols, DCF and EDCF, limit capacity at approximately 37% of the peak data rate
    b. Scheduled TDMA protocols such as HCF can theoretically reach around 90% capacity of the network, but under full load they will typically carry only approximately 75% of capacity
    c. DCF/EDCF MAC protocols do not effectively manage network latencies as the capacity limit is approached
    d. HCF protocols control latencies by providing fair weighted queuing so that all users will receive service even under full load conditions

The following table shows the throughput rates for HCF and DCF/EDCF for various modulations. These values can be de-rated when applied to larger cells that operate with lower capacity modulations.

	Throughput (MBPS)	
Modulation	HCF (75%)	DCF/EDCF (37
54 Mbps OFDM	40.5	19.98
22 Mbps PBCC	16.5	8.14
11 Mbps CCK	8.25	4.07
5.5 Mbps CCK	4.125	2.035

By and large, network designers do not use theoretical peak performance rates when planning a WLAN. As a rule of thumb, most network planners de-rate the theoretical performance figures to approximately 70% to 80% of the peak capacity.

Note: With packet aggregation and proper use of 802.11 protection mechanisms, DCF/EDCF can achieve higher levels of throughput (approximately 50% to 55% higher) with a limited number of users and limited number of connections requiring QoS capabilities. This does not address the concern many enterprise WLAN designers have for the stability of DCF/EDCF under a high user load.

Enterprise Capacity Analysis

Because an enterprise 802.11 WLAN deployment will involve covering a workplace with a series of APs, the network planner must analyze the bandwidth capacity of each cell and the bandwidth demands that users will make on each cell in the network. In an enterprise deployment, the APs will be connected to a router either directly or through an Ethernet switch. In larger enterprises, multiple sub-nets may be connected hierarchically so that a wireless subscriber actually passes through several routers before reaching the IP network.

This type of WLAN essentially represents a micro-cellular architecture using 802.11 APs interconnected via broadband IP links over Ethernet. APs have a certain coverage range which provides network access to users in a circular area around the location of the AP.

The analysis of enterprise network capacity that follows was based on the following assumptions:

- The average density of enterprise users is one per 200 square feet of floor space.
- The work day is eight hours long.
- 150 Mbytes of data as file downloads, e-mails and web accesses are transferred per user over the WLAN. No streaming media is supported.
- A sustained peak-to-average data throughput rate of three was used, essentially making the average data load three x 150 Mbytes or 450 Mbytes.
- Users require 0.15 ERLANG (15 me) of voice load. (This is based on current Bellcore and SBC business user peak busy hour loads.)
- A VoIP connection places a load on the WLAN of 64 Kbps in each direction (a combination G.726 and G.711).

Based on this profile, the following table illustrates the peak busy hour load on a WLAN cell as a function of the radius of the cell.

Cell radius (feet)	50	75	100	125
Users	39	88	157	245
Active Phone lines	12	22	34	49
Concentration X:1	3.25	4.00	4.62	5.00
Bandwidth (Mbps)				
Voice uplink	0.77	1.41	2.18	2.18
Voice downlink	0.77	1.41	2.18	2.18
Data downlink	3.25	7.33	13.08	20.42
Data uplink	1.63	3.67	6.54	10.21
Total Throughput	6.41	13.82	23.98	34.98

This network capacity analysis shows that even for a small cell with a radius of just 50 feet, a typical 802.11b network would not have the capacity for applications like VoIP or the "completely unwired workplace." However, if an 802.11a/802.11g WLAN with 54 Mbps modulation were combined with an HCF MAC in a cell with a 100-ft. radius, the cell would have nearly 40 percent reserve (excess) bandwidth. Alternately, if the inefficient EDCF MAC were used, a dual-mode 802.11a/g solution would be required to cover the same cell. Two RF channels would be required if the EDCF MAC were used.

"Wired When Docked" Workplace

The analysis presented above is based on the unrealistic assumption that users of a WLAN would always be completely wireless. In reality, a typical workplace will consist of wired and wireless users, and most wireless users will be "docked," or connected to a wired network, when they are at their desks.

Windows XP supports intelligent docking. Users are automatically switched from the WLAN network to a wired IP backbone when the device is docked. WLAN planners should take into consideration the effects that "wired when docked" will have on wireless networks' capacity requirements. For example, fewer than 20 percent of TI's workforce are un-tethered wireless workers. This has a profound impact on WLAN capacity needs, as shown in the following table.

Cell radius (feet)	50	75	100	125
20% wireless	1.28	2.76	4.80	7.00
30% wireless	1.92	4.14	7.19	10.49
40% wireless	2.56	5.53	9.59	13.99

Based on this analysis, planners can conclude that an enterprise WLAN with a "wired when docked" strategy can be supported by 802.11a/b/g dual-frequency access points using either HCF or EDCF. In other words, a deployment of a "sea of simple Ethernet-powered access points" would be sufficient.

In order to fully utilize the bandwidth of an access point, co-channel and adjacent channel interference must be addressed. The following section will briefly address RF planning.

RF Frequency Planning for Enterprise Deployment

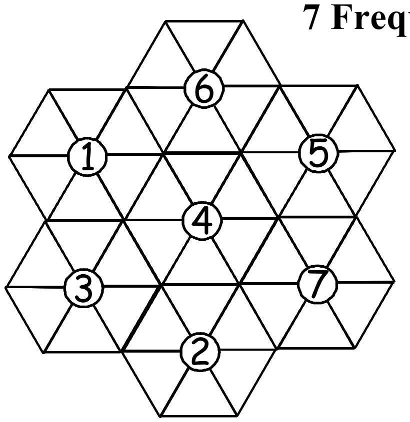
To analyze properly the overall capacity of a WLAN deployment, planners must consider the effects co-channel and adjacent channel interference will have on the throughput and bandwidth of the APs in the infrastructure. As WLAN APs are deployed for wide area coverage, WLAN RF interference issues take on characteristics similar to those that are faced in the planning of micro-cellular RF networks.

RF network planning begins with a consideration of the frequencies that are available. 802.11 a/b/g radios have the following independent frequencies:

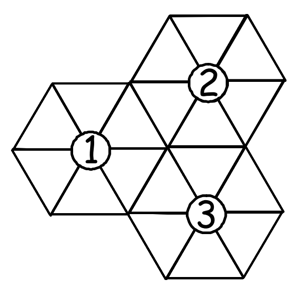
- 5.1 to 5.3 GHz with eight frequencies
- 2.4 GHz with three frequencies (There is some discussion in the industry that four frequencies actually could be used.)

For access points that are based on simple omni-directional antenna configurations, the following diagram illustrates both the seven-frequency and the three-frequency repeat patterns with frequency reuse of one. The seven-frequency plan can be used for 5.x GHz 802.11a, and the three-frequency plan can be used for 802.11b/g systems.

7 Frequency, Reuse = 1



3 Frequency, Reuse = 1



For these types of deployments, the cell reuse distance, R_u, can be defined as follows:

- C = 7 (7 frequency): $R_u = R_{cell}$*sqrt (3C) = 4.48* R_{cell}
- C = 3 (3 frequency): $R_u = R_{cell}$ *sqrt (3C) = 3.00* R_{cell}

Where:

C is the cluster size, which is the number of frequencies used in the reuse pattern
R_u is the reuse radius of the cell cluster
R_{cell} is the radius of coverage of a single cell

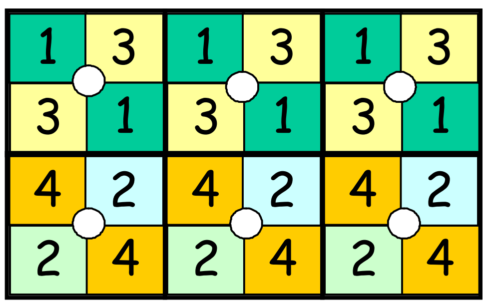
For distances greater than the AP's cell radius, it is assumed that RF propagation loss will not be free space (R^2) but will be R^3 to R^4. This would result in interference reductions between cells of at least the following:

- C = 7: 19.5 dB to 26.1 dB ***(allows 36 to 54 Mbps OFDM)***
- C = 3: 14.3 dB to 19.1 dB ***(allows 22 Mbps PBCC to 36 Mbps OFDM)***

Based on larger deployments, it would be possible to implement 802.11 a/b/g WLANs with omni-directional antenna coverage and allow automatic frequency selection at the access point so that the AP is able to establish the most effective frequency plan.

It is possible to use sectored access points and improve frequency reuse. However, in an enterprise environment this would require very careful placement of the APs and alignment of cell sectors. Where frequencies are at a premium, deployments based on four-frequency sectors per AP can provide optimal reuse. The interference reduction is equivalent to the omni-directional seven-frequency plan previously discussed. The following diagram illustrates the optimal four-frequency reuse plan.

4-Sector (smart) AP:
4-Frequency, Reuse = 1



Home Capacity Analysis

Unlike the enterprise market where some assumptions can be made about typical usage patterns, network capacity analysis for WLANs in the residential market will be greatly influenced by the rate of market penetration and the implementation of multimedia applications.

The following are some probable multimedia applications for the home:

- 802.11 VoIP cordless phones and home PBX/voice mail integrated into an 802.11 access point
- Streaming audio distribution to 802.11 speaker systems
  - Home PC as an MP3 audio service
- Streaming video from a cable television network, DVD system, etc.
- Telemetry applications, such as:
  - 802.11-enabled cameras/video for security
  - Meter reading for utilities
  - Smart appliances
- Wireless print server connections

If none of these applications are in demand by residential consumers in the near term, 802.11b with security features and QoS enhancements (802.11e/i) will meet the needs of most consumers. (Note: For consumers, speed will always sell. The concept that “faster is better” is compelling. For this reason, dual-mode 802.11b/g devices will have strong market acceptance as long as devices are backwards compatible with the nearly 20 million 802.11b subscriber base.)

Considering that the FCC has mandated that all TVs sold in the US must have a digital tuner, there is a very strong possibility of some level of wireless video distribution in the home. Video applications certainly will have the largest effect on the throughput and capacity requirements of home WLANs. The following table lists the bandwidth requirements for a number of current video codecs:

Video Media	Bandwidth (Mbps)	Packet size	Packets/sec	Delivered Bandwidth (Mbps)	Overhead (%)
MPEG-1 Basic Media	2.5	1,500	2083	2.5663	2.58%
MPEG-2 SDTV format (DVD)	8	1,500	6666	8.2125	2.59%
HDTV MPEG-2 committee 18	19.2	1,500	16000	19.7120	2.60%

This data indicates that a single MPEG2 SDTV/DVD quality channel requiring eight Mbps of bandwidth cannot be supported by current 802.11b MAC/PHY components. Fortunately, advances in video compression (MPEG-4) should reduce the bandwidth requirements for video applications to approximately one Mbps for DVD-quality video and about eight Mbps for HDTV-quality applications.

Over the next two to three years, many in the industry expect that a typical broadband-enabled household could have WLAN peak capacity needs as indicated in the table below:

Service	Rate Upstream Mbps	Rate Downstream	Number of Channels	Total Rate Upstream	Total Rate Downstream
MPEG DVD-TV	0.5	8	2	1	16
Toll Quality Voice	0.064	0.064	2	0.128	0.128
Streaming Media	0.01875	0.3	2	0.0375	0.6
ABR Web Service	0.0965	0.386	1	0.0965	0.386
			TOTAL	1.262	17.114

As shown in the following table, the market acceptance of HDTV and the absence of MPEG-4 compression could increase a home's WLAN throughput needs by a factor of four over the next four to five years.

Service	Rate Upstream Mbps	Rate Downstream	Number of Channels	Total Rate Upstream	Total Rate Downstream
HDTV	1.5625	25	2	3.125	50
Toll Quality Voice	0.064	0.064	4	0.256	0.256
Streaming Media	0.01875	0.3	1	0.01875	0.3
ABR Web Service	0.0965	0.386	2	0.193	0.772
			TOTAL	3.59275	51.328

These numbers indicate that high-throughput 802.11g/a PHY technology will be needed as a minimum in order to support these applications. Further, an efficient MAC (HCF) will be needed to optimize throughput.

VoIP applications do not require a significant amount of bandwidth in any of these capacity scenarios. Given the small number of phones in a typical home, the system must be designed for 1:1 concentration (that is, there would be no over-subscription of phone lines in the home). The more important benefits of WLAN-enabled cordless phones are twofold:

1. Removing cordless phones as a source of RF interference in the 2.4 GHz and 5.2-5.8 GHz frequencies could accelerate the acceptance of video applications over WLANs.
2. A new market for 802.11 cordless phones would be created with a sales potential of approximately 100 million units a year.

Residential 802.11 Link Asymmetry

Usage models of residential applications show that the typical data transfer load is very asymmetric. That is, the downlink from the AP to the subscriber usually requires 10 times more throughput than the up-link from the subscriber to the AP.

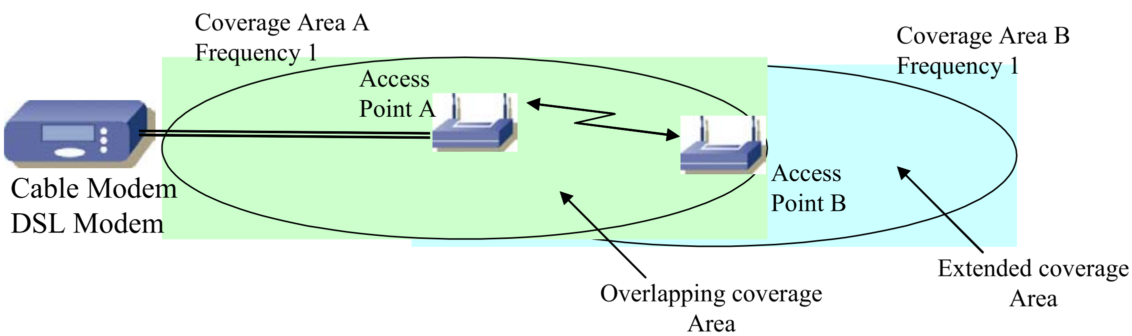
Radio architectures for 802.11 APs can be differentiated to improve coverage and throughput by simply "bolting on" a booster LNA and PA capability (much like an "afterburner"). This capability is most appropriate in North America where spectrum rules allow 10 dB greater EIRP (power) than in Europe.

The range of an AP can be nearly doubled at the highest modulation rate with simple link budget improvements in the AP.

Application of Repeaters/Small-mesh Access Points for Residential/SOHO Coverage

For developers of WLAN access points for the residential/SOHO marketplace, cell coverage and throughput are the most crucial issues facing WLAN implementations in this market. Wireless repeaters, which can be used to implement small mesh residential networks, are a low-cost method of improving coverage and throughput.

One possible technique for extending coverage and improving residential service is the use of multiple APs in a mesh/repeater architecture. A simple example featuring two access points is illustrated in the diagram below:



Access point B is a repeater (mesh element) to access point A, which connects to the Internet. Access point A functions as a router to access point B. Access point A must maintain a routing list for all clients in the home network while access point B only must maintain a routing list of attached clients. For example, B may be a simple bridge or a more intelligent router. Clearly, the mobility/roaming between the cells in this sort of arrangement will generate overhead messaging to update and maintain the routing information.

A real-world example of mesh WLAN architecture was the Aironet system, which was one of the first large-scale deployment platforms for WLANs. In this system, a client would probe for APs that could provide coverage, and the APs would reply with information on signal quality and on how much of their resources were currently in use. The subscriber would then associate and authenticate with the AP with the best signal quality and lowest usage. Once this was completed, re-routing updates would be completed.

Mesh networks can be nested deeper than a single connection. This is known as multi-hop. However, this creates even greater delays because of the cumulative time needed to route and retransmit from one AP to another. For voice, video phone and video conferencing, the round trip delay would be excessive for any architecture with more than one hop.

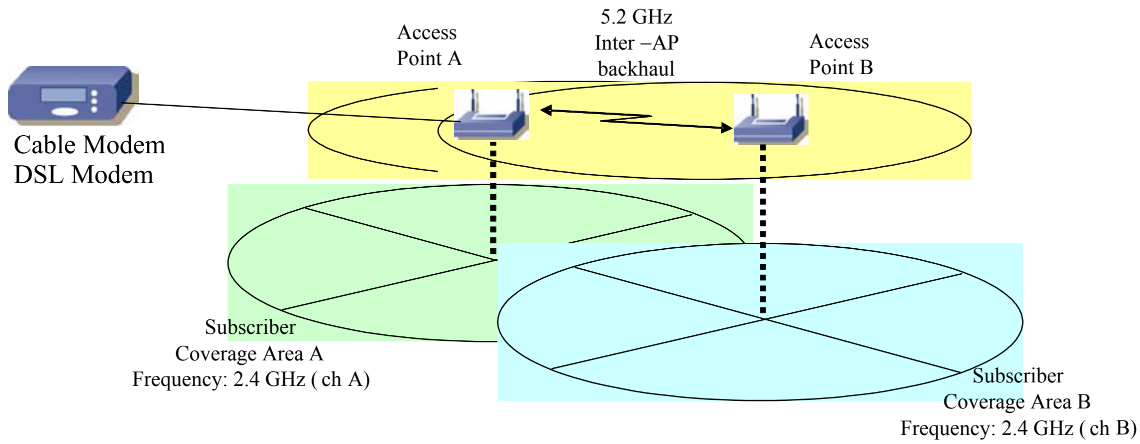
There are two possibilities for operating a residential mesh network. They are the following:

- **Single-Frequency Mode:** Access points are not dual-mode and can only support a single frequency of operation from an AP to another AP and from an AP to a client/subscriber.
- **Dual-Frequency Mode:** Access points are dual frequency, supporting two separate links on two separate frequencies simultaneously.

The single frequency mode of operation is backwards compatible to older single-frequency APs, but it is highly inefficient because the coverage provided by all APs in a WLAN is overlapping. Any communication initiated by an AP or a subscriber can interfere with any other communication. Under worst case conditions, the throughput is reduced by 1/(N+1) where N is the number of repeater/mesh APs attached to an AP.

The dual-frequency mode requires that all access points support two frequencies simultaneously. Typically, 802.11a (5.x GHz) would be used for AP-to-AP backbone communications while AP-to-subscriber communication would be provided by 802.11b/g (2.4 GHz). Using the 2.4 GHz frequency for subscriber coverage ensures support for low-cost and legacy 802.11b clients/subscribers. Because three independent 802.11b/g frequencies are available in the 2.4 GHz band, WLANs designed with a primary AP and one or two repeater APs actually improve the coverage of the home. Stated another way, as long as three APs are implemented, the coverage area is greater and throughput will be consistently high without RF interference between the APs.

The dual frequency configuration is shown in the following diagram:



Mesh/Micro-Cell and the Interference Environment

The IEEE community is debating whether to use MIMO and/or beam steering techniques for next-generation 802.11 standards as a way to improve throughput and coverage.

A simple mesh extension for the 802.11g standard combined with improved video compression could be available to consumers immediately, and this would provide a "virtual" performance improvement. Final approval of IEEE 802.11 HTSG is at least three years away.

The mesh/repeater architecture has another benefit in that it improves signal-to-interference (S/I) performance because the architecture ensures subscribers are consistently closer to access points. This, in turn, ensures better link margins.

Network Interfaces, Architectures and Timing Issues

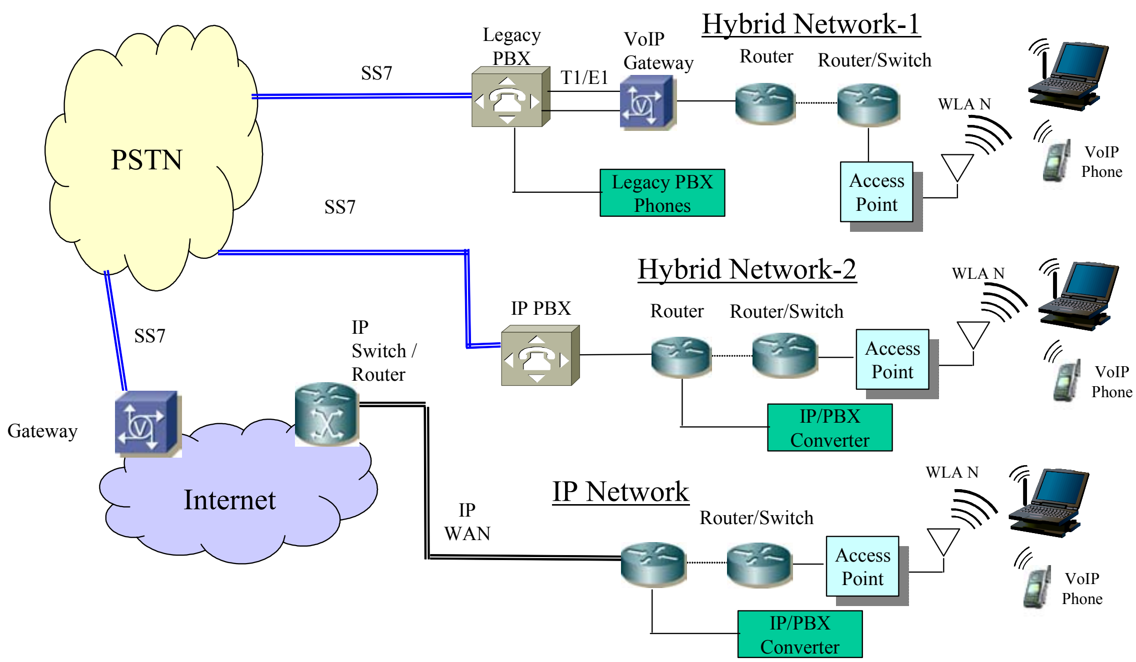
This section reviews the requirements of the PSTN with regards to a VoIP application as well as the timing issues that are critical for toll quality voice deployment.

How VoIP over WLAN applications will be deployed will have an effect on the design and integration of the equipment. The following issues have a bearing on equipment design:

- VoIP voice compression algorithm(s)
- Voice packet size, packet rate and delay
- Timing requirements for signaling and call set up
- Call control protocol
- Capacity and range of QoS capabilities that will be supported beyond voice

The market can be roughly divided between residential/SOHO and enterprise deployments.

The following figure illustrates enterprise deployments:



Enterprise Deployment Scenarios

VoIP integration in the enterprise will be evolutionary, not revolutionary. Greater than 90 percent of enterprises use analog and ISDN (P-Phone)-based PBX equipment. In some cases the PBX equipment is modular and supports VoIP interface LRUs (line replaceable units). For older legacy PBX equipment, a VoIP gateway can introduce VoIP into the enterprise.

Given the expense of analog PBXs and phones, enterprises typically decide to gradually transition the deployment of VoIP. The enterprise deployment diagram above illustrates a phased deployment such as the following:

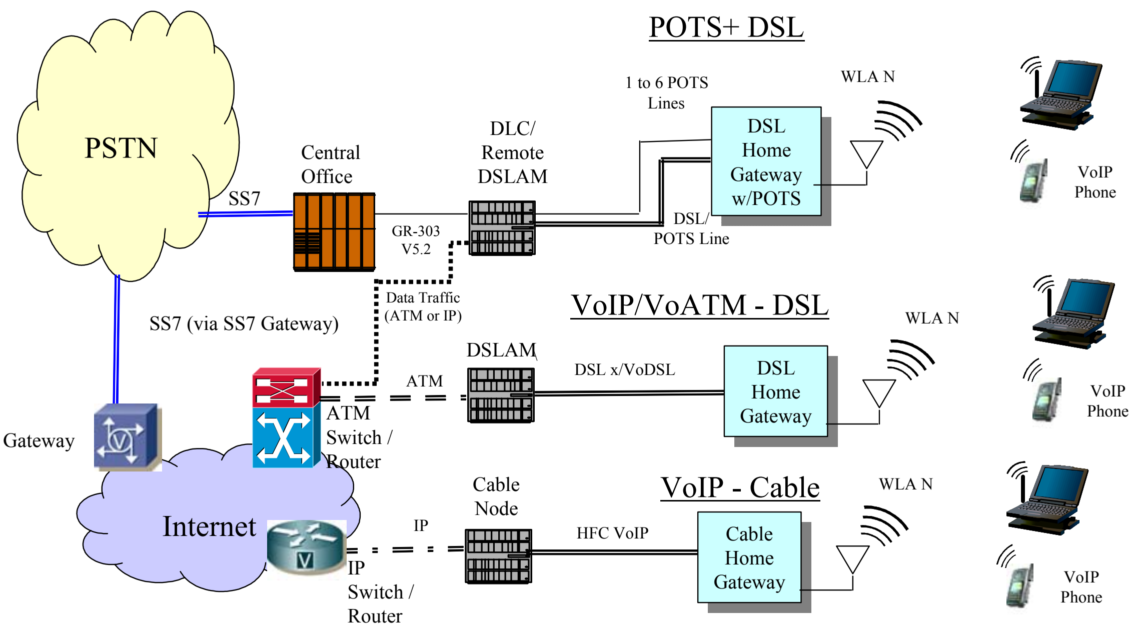
1. The analog PBX/voice mail system connects to the PSTN. A small number of users have VoIP phones connected via a VoIP gateway. (See Hybrid Network-1.)

2. A VoIP PBX/gateway connects the enterprise to the PSTN. A majority of users have VoIP phones. Legacy phones are supported by an IP/PBX converter. This network will be typical in larger enterprises with a large number of PSTN connections. (See Hybrid Network-2.)

3. The enterprise has an IP connection to a remote gateway/PBX that may serve one or more business customers. Legacy analog phones, if any, are supported by an IP/PBX converter. This network will be typical in small and medium size enterprises and in the remote facilities of large enterprises. (See IP Network.)

The enterprise solution is highly dependent on the data network topology. A large number of routers and the type of hierarchy among the routers could delay the network's throughput. The transition through each router will add to the delay budget for IP packets. Additional delay in the network could be caused by centralized security and authentication servers. Depending on the network topology, 802.11 clients who are roaming between APs may experience extended delays in accessing centralized authentication servers as well as longer latencies in completing handoffs between APs.

A residential network topology, as shown in the following diagram, will have its own set of challenges.



Residential/SOHO customers typically have one or more analog voice circuits which are connected directly to the PSTN through a CLASS 5 switch or through a DLC remote terminal. In many cases, subscribers are served by advanced DLC/DSLAM (DSL access multiplexer) remote terminals that provide a combination of DSL and POTS service on a single line as well as standard POTS voice circuits.

As the PSTN network has been upgraded to support DSL, integrated voice/data service (VoDSL) using ATM AAL1/2 packet voice has become available. In general, most residential and SOHO users will continue to have POTS as the primary voice interface. The following are several of the interface configurations that will emerge for home users:

1. POTS voice interfaces to the PSTN in a way that is identical to a cordless phone. VoIP conversion to/from the analog wire pair would be integrated into the VoIP-enabled access point.
2. VoDSL interfaces to the PSTN and VoATM is converted to VoIP for use with a WLAN AP.

Integrated DSL/WLAN gateways will become a trend because these gateways will be able to serve both of these residential/SOHO configurations.

If the residential/SOHO user is interfacing to a broadband cable modem, the topology is nearly identical to that of DSL. Where voice over cable is available, it is typically an independent system from the DOCSIS™ cable modem interface. An example of this independent interface is the ARRIS system, which results in termination of two-wire analog POTS interfaces at the customer premise. As DOCSIS 1.1 and 2.0 become readily available, integrated VoIP over cable modem will become prevalent.

The time delay of the communications path for 802.11 VoIP in a cordless residential application will typically be much shorter and have a limited number of sources of delay. The voice interface to the home will typically be a POTS interface. Specific signaling requirements must be supported with analog line pairs to DLC and/or CLASS 5 switching equipment. (See note below.)

WLAN Lower Sleep Modes and POTS Call Processing

The design of DLC and Class 5 switch equipment is based on the premise that the wire connection to the phone is in place and operational. Typically, establishing a call with Q.931 protocol is completed in less than 200 msec. The Caller ID modulation receiver (FSK Modem) must be ready after the first ring, which is less than or equal to two seconds. Worst case signaling delay should be less than 100 msec.

This creates a problem.

Any WLAN implementation will include VoIP handsets that are often in power-saving sleep modes where much of the device is not operating. In this mode, the WLAN will “wake up” and establish communication in intervals of 200 msec to one sec, well beyond the telephony system delay specification.

Clearly, the 802.11 AP will have to maintain an attached state for a given handset and provide a call proxy during call set up until the call can be handed off to the handset after it has awakened from sleep mode.

Access Time Delay

Regardless of the application, the time delay and jitter of the VoIP system will be a design consideration. As already noted, the two following issues relate to time delay and jitter:

1. Signaling for call set up, tear down and other call control communications will be delayed. (Worst case delay is the principal concern.)

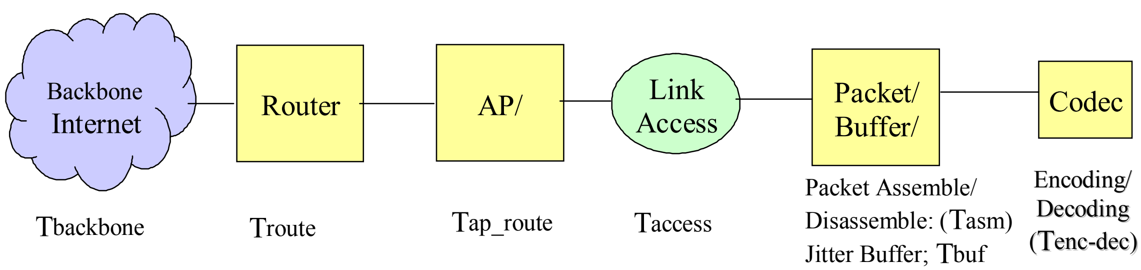
2. Jitter in the voice traffic/bearer channel will cause delay.

VoIP signaling and voice traffic are not separate communications channels. VoIP packets exist as virtual communications within a single channel. Only queuing priorities can ensure timely delivery of voice packets when other types of packets are competing for services in an IP network. This situation is complicated when a wireless user is moving and there are AP-to-AP handoffs in the network. Further delays are added into the WLAN as the user must associate with an AP, authorization must take place and the handoff must be completed.

The ITU has set recommendations for the maximum round trip delay in a voice system and the perceived quality of the voice channel. This recommendation is defined in ITU G.131 and is provided in the following table:

G.131 Delay	
0 to 150	acceptable to most
150 to 400	acceptable for international
> 400	unacceptable for public network

Under normal operations, the roundtrip delay should be less than 150 msec. The following diagram illustrates the possible delays in the communication path for an enterprise network with one layer of routing.



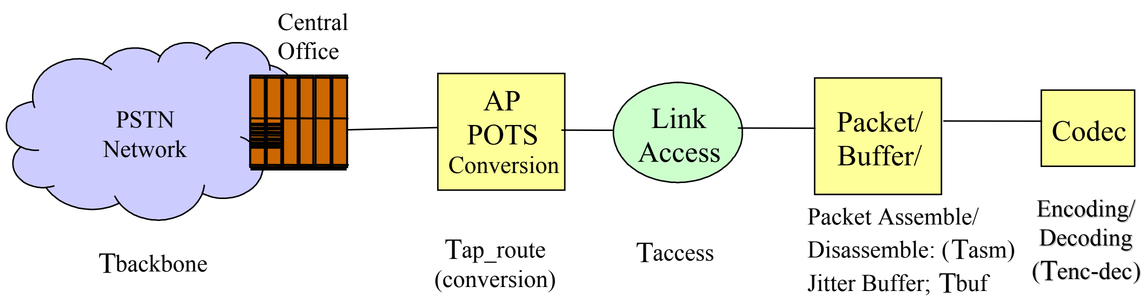
Because delay in the PSTN backbone network is beyond the control of mobile device manufacturers, the following table illustrates a residual delay budget for the backbone network (Tbackbone) using short duration G.711 and G.726 VoIP packet structures. The data for this table is taken from measurements that were performed on TI/Telogy VoIP software implementations and from worst case delays in TI's extensive wireless enterprise network.

Round Trip Delay Source	Delay Symbol	G.711 (5 msec)	G.726 (10 msec)
Encode/Decode	Tenc_dec	5.5	5.5
Assemble/Disassemble	Tasm	10	20
Jitter Buffer (1)	Tbuf	10	20
WLAN Access Delay	Taccess	10	20
Access Point Routing	Tap_route	10	10
Enterprise Routing	Trouting	5	5
Backbone Delay (residual)	Tbackbone	**99.5**	**69.5**
	Total	**150**	**150**

(1) Tbuf can be set as large as Tbackbone

These figures indicate that the G.726 residual backbone delay will be less than 70 msec, but there is some concern in the industry that in countries with large geographic areas, such as the U.S., Canada and others, the backbone network delay may exceed 70 msec.

For a VoIP cordless phone, the equivalent diagram of the communication path would remove the enterprise router and feature a minimum of 10 msec of additional delay margin (the maximum delay for the POTS PSTN network). The residential delay path is shown in the following diagram:



Any effects communication delay could have on voice quality for a fixed implementation of VoIP over WLAN is readily ameliorated, but larger issues arise when a user is handed off from one AP to another.

In a cellular phone system, a great deal of effort is expended on the handoff operation. The handoff is typically completed in 35 msec with 50 msec being worst case. WLAN systems do not have the interconnect processing capabilities or higher-order switching intelligence that is built into cellular networks. In a WLAN, the following capabilities are relevant to the network's ability to hand off active phone calls:

1. The WLAN must know when a link has been lost. (This can be a simple rule, such as losing more than N/M packets.)
2. AP probe and associate.
    a. Currently, tests by the University of Maryland show that an AP probe takes place in **250 to 400 msec**.
    b. Significant effort to improve both AP-to-AP handoffs and authentication are being addressed by the IEEE 802.11 committee's 802.11i (security) and 802.11e (QoS) task groups.
3. Authentication, security and routing updates.
    a. Delays of more than three seconds have been caused by centralized authentication servers.

Clearly, dramatic improvements are needed in AP probe, authentication and routing update operations. WLANs have delays for handoffs nearly 10 times greater than handoffs in cellular systems.

During the handoff between 802.11 APs, there will be short but noticeable loss of voice packets. On the positive side, VPN and call agent servers have timeouts on the order of tens of seconds. While some VoIP packets may be lost in a WLAN, connection should be maintained.

Several proposals for solving the hand-off delay problem have been proposed. These include the following:

1. Nearest neighbor or sub-net authorization proxy (authorization across a sub-net).
2. Highest QoS priority to an AP's probing services.
3. Shadow registration in enterprise WLANs. That is, a subscriber will be pre-registered and authorized within a sub-net.

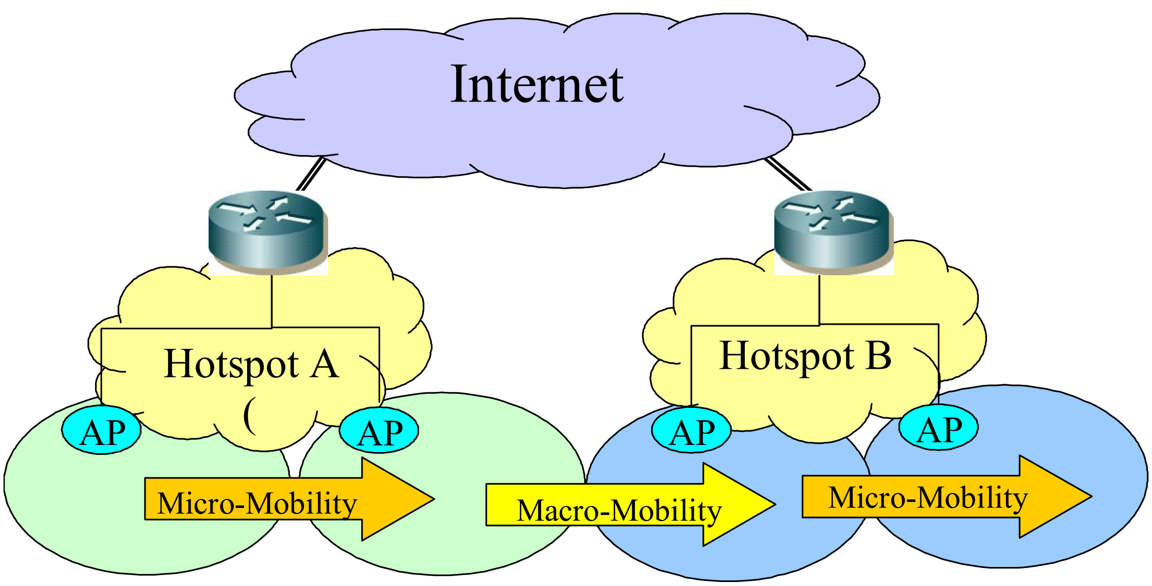
Work in these areas will continue for quite some time.

VoIP Protocols and Wide Area Networks

With the large scale rollout of wireless Ethernet-based VoIP phones in the enterprise, VoIP technology has reached a level of maturity where residential VoIP cordless phones, wireless PBXs and, eventually, future cellular systems based on 802.11 are no longer just a possibility for VoIP over WLAN technology. They are inevitable.

The larger challenge of VoIP over WLANs will be how to handle handoffs of an active call between 802.11 APs or between an 802.11 AP and a cellular network's cell. VoIP residential cordless phones will be the only application that will not require handoff capabilities initially.

The following diagram illustrates the basic mobility challenge for WLAN implementations:



As the diagram illustrates, the hierarchical nature of IP network topology results in two types of mobility:

- Micro-Mobility
- Macro-Mobility

Micro-mobility and macro-mobility are defined as changes of access point association (attachment) while an on-going VoIP (or data) session is in progress. They define the requirements for handoffs in the larger system. Micro- and macro-mobility differ from WLAN roaming or nomadic operations where a session is simply terminated and restarted in a new 802.11 AP cell. (This is what happens in WLAN hotspots today.)

Micro-mobility is the simplest form of mobility. The subscriber is moving within a single domain, such as an enterprise, a set of hotspots owned by company A or some other sort of limited WLAN configuration. Micro-mobility essentially involves intra-domain handoffs. There is no need for external coordination. Issues of timing, call control and handoff control can be set (or bounded) by network design. The first wave of VoIP over WLAN services will be based on micro-mobility in the enterprise (e.g. Cisco CCx effort) or in the residence (e.g. TI's solution for mesh/repeaters).

Macro-mobility involves moving between two domains that fall under the administration of completely distinct organizations. For example, one hotspot could be run by carrier A and a second is administered by carrier B. The two domains must collaborate to complete the handoff and to conduct authentication, authorization and accounting (AAA) activities between the domains. These arrangements are similar to efforts in the cellular industry that have been developed over the last few years.

Given that micro-mobility solutions will be the first developed and deployed, micro-mobility solutions must consider the larger framework that includes macro-mobility capabilities as well as the eventual evolution to full macro-mobility.

There are two principal approaches for supporting mobility in VoIP services. They are:

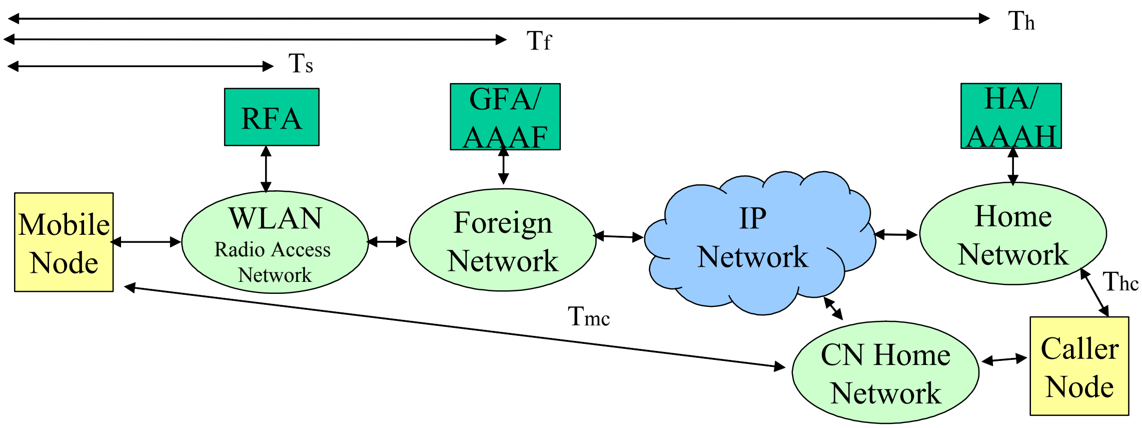
- SIP (Session Initiated Protocol)
- Mobile IP

Mobile IP is a network layer (i.e. layer 3) approach to mobility. While Mobile IP does not directly support VoIP applications, the protocol can be used as a basis for VoIP with additional and potentially proprietary protocols (e.g. Cisco's CCx). The alternate solution is to confront the mobility challenge at the application layer (layer 4/5) by augmenting existing VoIP protocols like SIP or H.323. Currently, the inclusion of SIP in Microsoft Windows™ XP has resulted in widespread support and a proliferation of the infrastructure for the simpler SIP protocol over the more rigid H.323 protocol.

The following section gives a brief overview of Mobile IP and SIP for mobility applications and concludes with a discussion of cellular GPRS/WLAN integration for data services. Many in the industry anticipate that cellular data deployments like GPRS/WLAN will be implemented initially and a larger movement to full blown VoIP WLANs will follow later.

Mobile IP Overview

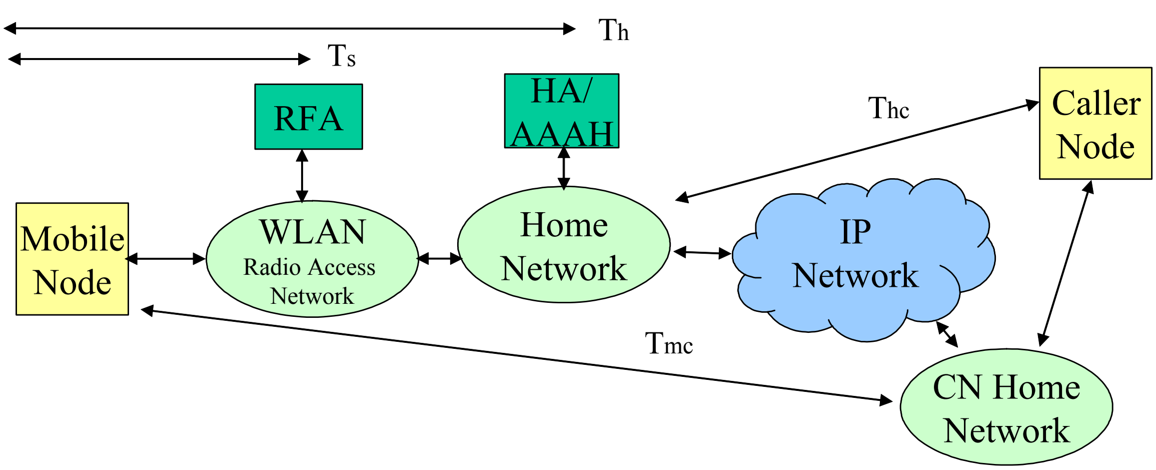
Several elements are needed to implement a WLAN incorporating Mobile IP capabilities for AP-to-AP handoffs. The following diagram illustrates the elements of a Mobile-IP system:



The elements shown above are defined as follows:

- Mobile Node: VoIP caller
- RFA: Regional Foreign Agent
- AAAF: Authentication, Authorization, Accounting - Foreign Network
- GFA: Gateway Foreign Agent
- HA: Home Agent
- AAAH: Authentication, Authorization, Accounting - Home Network
- CN: Corresponding Node (the network that a caller node is attached to)
- Caller Node: Another phone caller

In an enterprise network using micro-mobility, the foreign agent is removed and replaced by the home agent. This is shown in the following diagram:



When both caller and the mobile node are in the home network (i.e. when the switching occurs within one's own voice network), the PBX function is present. In this case, the VoIP call is routed through a self-contained enterprise network and no Internet domain resources are required.

Mobile IP is based on the concept that a mobile node has a home address associated with a home network. Each time the mobile node connects to a foreign network, it obtains a temporary address which is known as a Care of Address (CoA). The CoA is valid while the mobile node is attached to the foreign network domain. It is deleted or purged from the foreign network once the mobile node leaves the domain.

In Mobile IP WLANs, there are two mobility agents, Home Agents (HA) and Foreign Agents (FA), that coordinate, update and authorize the connections and associated CoAs for clients from foreign networks. When a call is set up between a Caller Node and the Mobile Node, a binding update message is sent by the Home Agent to the Corresponding Node. The binding message allows VoIP traffic and messages to be directly tunneled between the Caller Node and the Mobile Node. Messages need not be routed to the home network. Clearly, tunneling greatly increases the efficiency of Mobile IP.

Difficulty arises when roaming occurs between two foreign networks while a call is in progress between a Caller Node and a Mobile Node. Consider a Mobile Node that is leaving one foreign network and transitioning to a new foreign network. The first foreign network and associated FA must send binding warning messages to the caller's Home Agent (HA), alerting the HA that packets from the CN are arriving, but the Mobile Node is no longer in the first foreign network. The first foreign network redirects any received packets back to the caller's HA. Simultaneously, the Mobile Node will complete the association with the second foreign network. The HA then will establish the new CoA with the second Foreign Agent / foreign network. The HA then provides a new CoA to the second foreign network, and this allows direct tunneling of messages between the Mobile Node on the second foreign network and the caller node in the corresponding network. While this handoff is taking place, the HA acts as a middle man, maintaining the VoIP connection by forwarding packets from the first foreign network to the second foreign network. This stops once the new CoA and binding for direct tunneling are in place.

A complete description of this process is beyond the scope of this paper. However, it is obvious that when the problem is reduced to a single domain (such as an enterprise WLAN where there are no foreign networks), the solution is much more tractable.

In addition to the overall handoff process, controlling several timing issues is imperative for successful macro-mobility and micro-mobility handoffs. The following are some of the timing elements that must concern WLAN or handset designers:

- T_s: The period of time needed for a station to associate with an access point (probe and associate)
- T_f: The period of time needed by a handset to associate with a foreign network (inter-domain update)
- T_h: The period of time to bind to a foreign network and create a new CoA
- T_{mc}: The period of time needed to send packets directly between the Mobile Node and a Caller Node
- T_{no}: The period of time needed to bind update messages from an old foreign network to a new foreign network

The time required to register and set up a VoIP call for Mobile IP is:

$$T_{mip\_init} = 2T_s + 2T_h + 2T_{mc}$$

For macro-mobility (inter-domain) handoffs between two different foreign networks, Mobile IP has the following timing:

$$T_{mip\_inter} = T_{no} + 3T_h + T_{hc} + T_{mc}$$

During a handoff, as the new tunnel connection is established between a Mobile Node and the Caller Node, a series of packets will be disrupted and may arrive out of order, causing them to be discarded. The period of time for this disruption is given by the same formula for both SIP and Mobil IP. It is:

$$T_{black\_out} = 2\ T_s + 2\ T_h + 2\ T_{no}$$

For micro-mobility handoffs in the same domain, such as a handoff from one AP to another in an enterprise WLAN, Mobile IP has the following timing:

$$T_{mip\_intra} = 2T_s + 2T_f$$

At this point, issues relating to intra-domain handoffs will be discussed. In the next section, the SIP approach and its associated timing issues will be described.

Issues for Mobile IP Macro-mobility

There are two major issues relating to the implementation of Mobile IP macro-mobility:

- The probe and association time for 802.11 APs is not included in the time delay budgets in the previous discussion. This will exacerbate handoff delays unless improvements are made.

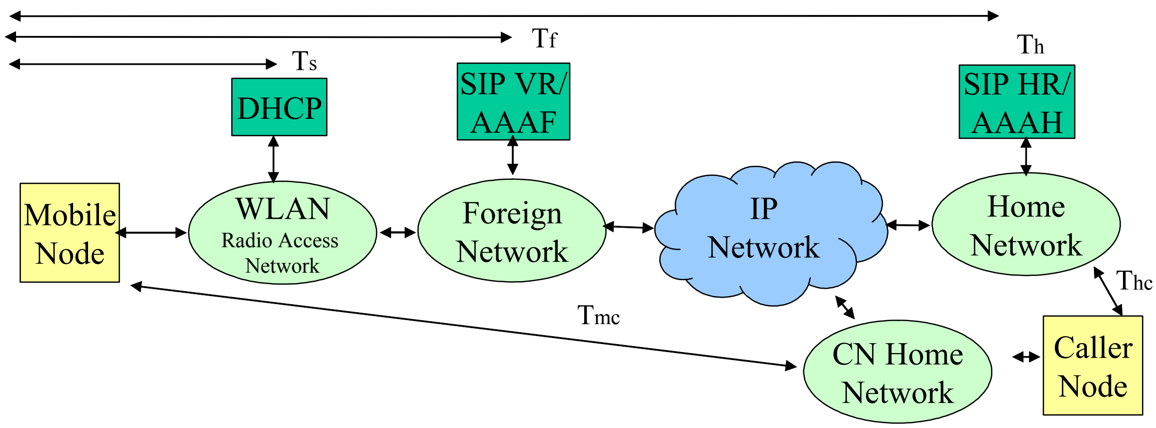
Lost packets may result in short disruption in voice services.
A more important issue is the fact that Mobile IP is not widely supported. The end-to-end deployment of Mobile IP on the Internet will take significant effort to achieve.

SIP Overview

As an alternative to Mobile IP, SIP supports IP mobility for VoIP WLAN applications by providing handoff capabilities at the application layer.

SIP can make direct use of Dynamic Host Control Protocol (DHCP) when connecting to an 802.11 AP for binding an IP address. A number of proposed systems use DIAMETER as the AAA (authentication, authorization, accounting) protocol. SIP makes use of the concept of a visited registrar (VR) in the foreign network. The SIP VR combines some of the functions of an SIP proxy server, location server and user agent. The SIP proxy server concept allows SIP to handle both firewall functions and network address translations (NAT), which are pervasive in home network topologies. SIP was initially designed to support roaming (i.e. moving into a domain while the connection is disabled and then establishing service) so that a user could be found independently of location and network device. For example, with SIP a call on a handheld phone could be transferred to a computer SIP phone. SIP is being modified to support mobility as well as roaming applications.

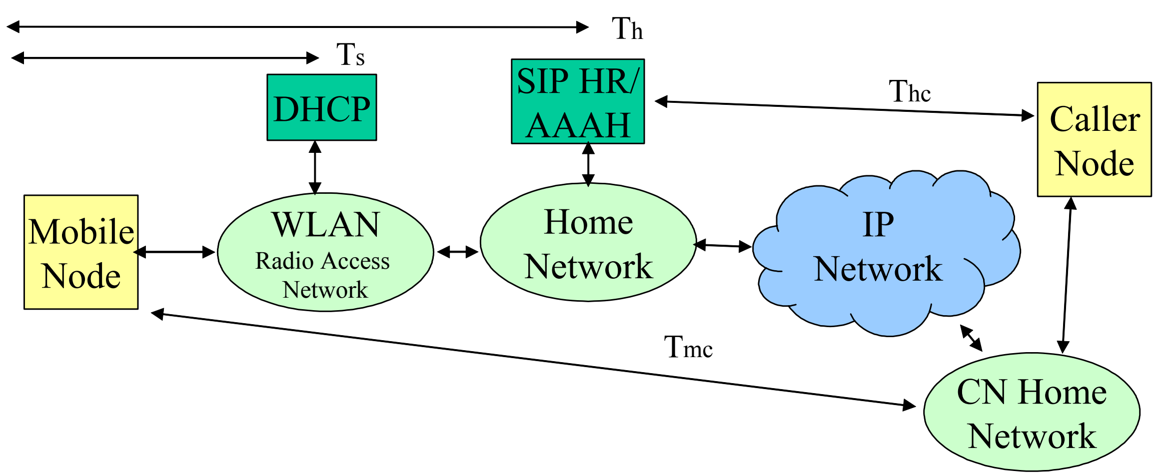
Like Mobile IP, macro-mobility in an SIP implementation would be based on the concept of foreign networks and home networks. With SIP, the foreign agent of Mobile IP is replaced by an SIP VR in the foreign network. The Mobile IP home agent (HA) is replaced by an SIP home register (HR). The SIP HR is a combination of an SIP proxy server, a location server and a user agent server. The following diagram illustrates an SIP network:



The elements in this type of network are defined as follows:

- Mobile Node: VoIP caller
- DHCP: Dynamic Host Control Protocol
- AAAF: Authentication, Authorization, Accounting - Foreign Network
- SIP VR: Visited Registrar
- SIP HR: Home Registrar
- AAAH: Authentication, Authorization, Accounting - Home Network
- CN: Corresponding Node (the network that a caller node is attached to)
- Caller Node: Another phone caller

When VoIP callers are in their home networks or a self-contained enterprise network that has implemented SIP-based micro-mobility, the VR (visited registrar) is removed and replaced by an HR (home registrar). This is shown in the following diagram:



One of the principal differences between Mobile IP and SIP is the use of DHCP by 802.11 APs. DHCP doubles the number of transactions needed to associate with an access point. It also requires that the client perform an ARP (Address Resolution Protocol) to detect duplicate addresses in the sub-net. The advantage of DHCP is that no modification to the local network is needed.

While there are minor differences between Mobile IP and SIP, the handoff procedures are essentially identical. SIP has the advantage of using the existing IP network without modification. However, this comes at the expense of delays that are typically double those of Mobile IP in a macro-mobility environment.

To its advantage, SIP is fully supported today by the Windows environment (i.e. Windows XP), making possible a rapid deployment in the residential/SOHO marketplace.

Just as with Mobile IP, timing issues must be addressed if SIP handoffs are to be supported. For the most part, SIP's timing elements are identical to those for Mobile IP. The only exception is SIP's use of the Address Resolution Protocol (ARP). In the formulas below, T_{arp} is defined as the period of time needed for an ARP exchange.

The time required to register and set up a VoIP call with SIP is the following:

$$T_{sip\_init} = 4T_s + T_{arp} + 2T_h + 2T_{mc}$$

(addition of $2T_s + T_{arp}$ vs. Mobile IP)

For macro-mobility (inter-domain) handoffs between two different foreign networks, SIP has the following timing:

$$T_{sip\_inter} = 4T_s + T_{arp} + 2T_h + 2T_{mc}$$

(For initially establishing service, SIP's timing is identical to Mobile IP, but much greater than T_{mip_inter})

The blackout time for SIP and Mobile IP is given by:

$$T_{black\_out} = 2\,T_s + 2\,T_h + 2\,T_{no}$$

For micro-mobility (intra-domain) handoffs such as an enterprise-based AP-to-AP handoff in the same domain, Mobile IP has the following timing:

$$T_{mip\_intra} = 4T_s + T_{arp} + 2T_f$$

(addition of $2T_s + T_{arp}$ vs. Mobile IP)

As these formulas indicate, by using the existing IP network without modification, SIP suffers from delays that can be 2x those of Mobile IP.

<u>SIP for Residential and SOHO Use:</u>

SIP can be used on an existing network without modification.

SIP is designed into Windows XP and will be in Windows CE.

SIP suffers from delays that can be **<u>2x</u>** those of Mobile IP. In a larger network, these delays quickly become unacceptable.

For cordless phone VoIP applications in the home, the delay in SIP is negligible and its ease-of-integration will greatly facilitate product introduction.

IPv6 and Protocol Improvement

These discussions of Mobile IP and SIP have assumed WLAN deployments on IPv4 networks. Both SIP and Mobile IP would greatly benefit from the pervasive use of IPv6. In both cases this would allow direct addressing of a mobile node client. If Mobile IP and SIP were revised in light of IPv6, the Foreign Agent could be removed entirely and handoff times would improve.

For Mobile IP to succeed, it must be deployed pervasively. As a result, there is interest in the industry in SIP for consumer applications.

Wide Area Network Integration: WLAN and GPRS Inter-working

While the ultimate goal is to provide seamless IP mobility for all applications including voice, in the near term cellular carriers are planning to integrate data operations through a combination of 802.11 for hotspots and cellular telephony technology for wide area data networking.

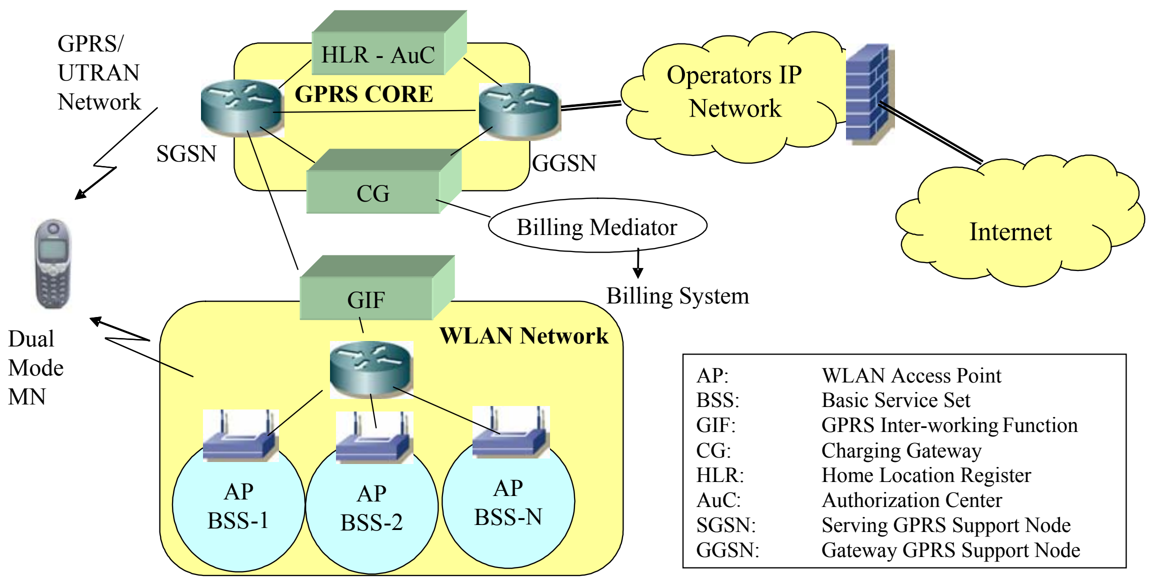
This section briefly describes the integration of the cellular GPRS (General Packet Radio System) standard with WLAN technology in a seamless data network. This process will take several steps, including the following:

1. Common billing and customer care but no inter-working of WLAN and GPRS networks.
2. A 3GPP-based access control and charging system where all WLAN AAA (authentication, authorization, accounting) will be based on GPRS AAA procedures.
3. Access to GPRS data services such as WAP are supported on the WLAN system, but there are no handoffs between WLAN and GPRS.
4. Where jitter and time delay permit, there would be service continuity for the services described in item three above. These services would be provided across the WLAN and GPRS networks. The handoff of IP multimedia may not be supported, but other IP services would be.
5. Seamless continuity where all services are supported transparently between WLAN and GPRS networks. There is no noticeable difference in the services.
6. Access to 3GPP switched circuit services is provided and voice services are supported.

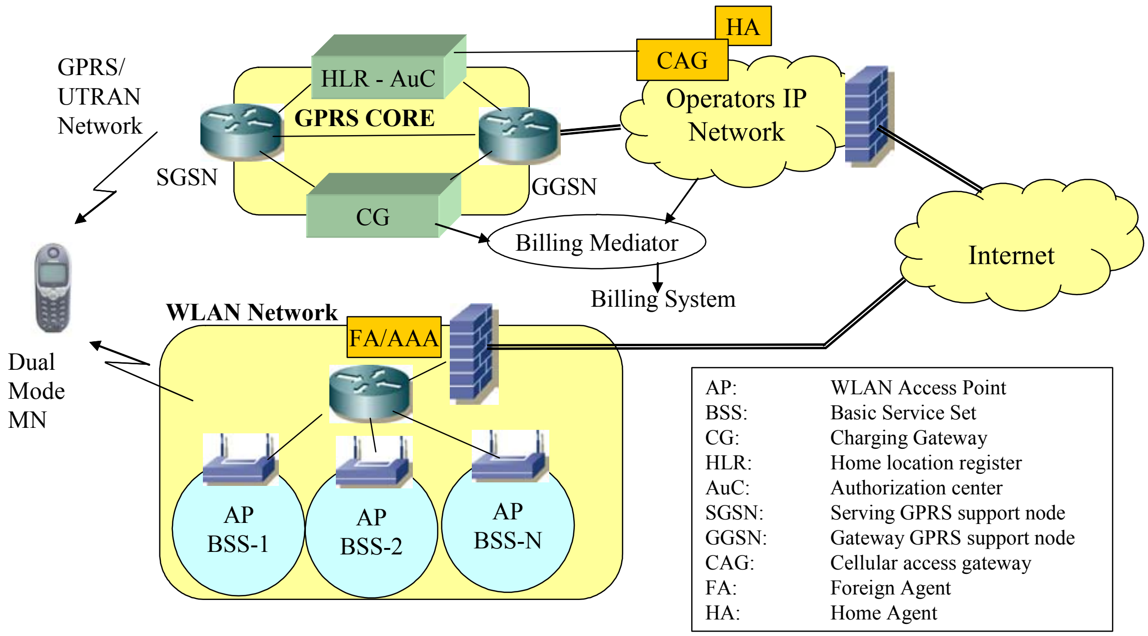
There are essentially two schools of thought on how WLANs should interface to the existing GPRS network. These views are:

- Tightly coupled WLAN network
- Loosely coupled WLAN network

A tightly coupled network is illustrated in the following diagram. The tightly coupled network makes use of all existing GPRS system resources for networking, AAA, security, provisioning and other functionality. These functions are coupled to the WLAN systems. With minor exceptions, the WLAN user will have immediate access to all GPRS services. This type of system would use a strong GPRS inter-working function (GIF) to interface to the WLAN network, and all traffic would be routed through the Serving GPRS Support Node (SGSN).



A loosely coupled network is illustrated in the following diagram. This loosely coupled scheme would be based on Mobile IP. Only minor modifications to installed WLAN networks would be required. However, cellular operators would need to install AAA servers for billing mediation and to support WLANs as well as to support interoperations between the GPRS network and WLANs. In a loosely coupled system, the Internet is used as the traffic backbone. Because service operators would not have complete control over the network, there is some concern that consistent quality might not be provided.



The following table lists the differences between tightly and loosely coupled systems.

Category	Tight Coupling	Loose Coupling
Authentication	GPRS authentication and cipher key encryption	SIM based authentication. Optional Radius based
Accounting	Reuse GPRS accounting	External Billing for common accounting
WLAN Cellular Mobility	SGSN Call Anchor, mobility by inter-SGSN handovers	Home Agent (HA) is the call anchor, mobile IP between access router and GGSN (gateway)
Context Transfer	Fine grain info on QoS, flows, etc.	Limited information between GGSN and WLAN (IETF working on "seamboy")
System/Network Engineering	Impact on WLAN traffic to existing GSN bearer and signaling is an issue	WLAN and GPRS can be designed separately
New Development	WLAN modifications for GPRS for GPRS signaling, possible SGSN modes	CAG for SIM-based authentication, Billing mediator for accounting
Standards	A new SSN interface	EAP-Sim and EAP-AKA authentication (IETF Ppext working group)
Target Usage	Applies to Cellular owned WLAN or affiated WISP	Broad Application

The deployment of WLAN hotspots has taken on a life of its own. It may not be practical to require conformance to certain standards or to modify existing WLAN equipment with cellular security and signaling. The motivation of carriers is clear. Carriers require control over network quality. They also are concerned that Mobile IP has not been widely deployed and, because of this, seamless interfaces may be delayed. Carriers also are concerned that when QoS capabilities are deployed, they may be haphazard at best. Clearly, all indications from the marketplace would suggest that the deployment of seamless networks of WLAN and cellular technology will happen in the near future. Indeed, certain cellular carriers already have acquired and are supporting WLAN hotspot networks, merging the billing and customer care operations for the two technologies.

Deploying VoIP over WLANs Today

As previously stated, the technology needed to deploy VoIP over WLANs and the other wireless applications described in this white paper exists today and is being incorporated into next-generation handsets, mobile devices, personal digital assistants (PDAs), laptop computers, infrastructure systems and other types of systems. In fact, TI's WANDA concept design is an apt example of how leading-edge wireless technology can be designed into advanced systems today.

WANDA, which stands for Wireless Any-Network Digital Assistant, is a handheld tri-band device that integrates 802.11 WLAN, GSM/GPRS and Bluetooth™ into a PDA concept design. WANDA features several of TI's industry-leading components, such as the OMAP1510 application processor, the low-power TNETW1100B 802.11b MAC/baseband processor, the BRF6100, the industry's first single-chip Bluetooth solution with digital RF, and the TCS2100 GSM/GPRS chipset. Despite its impressive processing capabilities, WANDA capitalizes on one of the most important priorities of wireless subscribers: extended talk time and standby time. Estimates indicate that WANDA is capable of 450 hours of GSM standby time, 12 hours of PDA constant usage time and eight hours of GSM talk time on a single battery charge.

Many of TI's other advanced wireless components are being deployed in next-generation wireless applications. Some of these devices are listed below:

Application Processors:

OMAP1610	Dual-core DSP/RISC for high-end multimedia.
OMAP1510	Dual-core DSP/RISC for PDAs, Pocket PCs, smartphones and others types of mobile devices.

Communication/Application Processors:

OMAP710	GSM/GPRS modem and application processing core.
OMAP730	GSM/GPRS modem and application processor for smartphones, PDAs and handsets.

WLAN Processors:

TNETW1100B	Single-chip 802.11b MAC/baseband processor for low-power mobile applications.
TNETW1130	Single-chip 802.11a/b/g MAC/baseband processor for multimode devices with WLAN throughput of 54 Mbps.

Bluetooth Processors:

BRF6100	Complete single-chip Bluetooth subsystem with digital RF.

Cellular Telephony Chipsets:

TCS2100	Full Class 12 GSM/GPRS solution with digital and analog baseband processors and RF transceiver.
TCS2600	In addition to full Class 12 GSM/GPRS capabilities, includes OMAP730 smartphone processor for accelerated application processing.

Conclusions

This white paper has shown that the technology exists today to implement VoIP over WLANs applications as an integral part of next-generation seamless wireless voice/data networks. The wireless industry has already begun to migrate toward an environment where one phone number can be used practically anywhere for voice and data applications. To accomplish this goal, mobile device designers, carriers, service providers and enterprise/home network designers face deployment, provisioning and implementation issues, many of which were described in this document.

In summary, some of this white paper's major conclusions are as follows:

- The data rates of current 802.11a/g MAC/baseband processors will support the throughput needs of VoIP over WLAN applications as well as other next-generation multimedia applications like video streaming.
- RF interference will be a factor in deploying next-generation wireless networks that include 802.11 WLAN capabilities.
- 802.11 AP-to-AP handoffs in VoIP over WLAN applications can be managed effectively with today's technology.

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