Implementing VoIP Service Over Wireless Network

BreezeACCESS® VL

White Paper

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Introduction

Voice over Internet Protocol (VoIP) technology facilitates packet based IP networks to carry digitized voice. With VoIP, competitive carriers and service providers can offer telephony (voice and fax) services together with traditional data services over the same IP infrastructure, and by doing so, increase revenue stream and improve business models.

Constructing a VoIP telephony service over a wireless IP network requires understanding of VoIP technology and the unique characteristics of the wireless medium to maximize call quality and capacity.

This document includes the following:

- VoIP Basics: Introduction to VoIP technology, description of most popular CODECs and definition of some important properties.
- When VoIP goes Wireless: Description of the main implications of using VoIP over wireless networks.
- Choosing the Right VoIP Gateway CPE: Describes the VoIP Gateway characteristics and capabilities that affect voice quality and cell capacity when using it over a wireless link.
- VoIP over BreezeACCESS VL: Describes the BreezeACCESS VL and its main capabilities, explains how to optimally configure the VoIP Gateways and the BreezeACCESS VL system for best performance. It includes also some examples and a case study.
- Summary

1 VoIP Basics

VoIP equipment uses Compression/Decompression (CODEC) technology for converting audio signals into a digital bit stream and vice versa. Over the years, compression techniques were developed allowing a reduction in the required bandwidth while preserving voice quality. Although many compression schemes exist, most VoIP devices today use CODECs that were standardized by bodies such as the ITU-T for the sake of interoperability across vendors. The most popular CODECs are:

- **G.711** is the basic, most common CODEC. G.711 CODECs use Pulse Code Modulation (PCM) of voice frequencies at the rate of 64 Kbps. It covers both "A-law" and "μ-law" encoding. A-law and μ-law are compounding schemes used to get more dynamics to the 8 bit samples that are available with linear coding. Typically 12..14 bit samples (linear scale) sampled at 8 KHz sample rate, are compounded to 8 bit (logarithmic scale) for transmission over a 64 Kbps data channel. In the receiving end the data is then converter back to linear scale (12..14 bit) and played back. The μ-law is used in North America and Japan, and the A-law is used in Europe and the rest of the world and in international routes. This non-compressing CODEC requires low computation complexity and provides very good voice quality with negligible delay. However, it consumes 64 Kbps per direction, which is high compared to other CODECs.
- **G.723** CODEC has two bit rates associated with it, 5.3 and 6.3 Kbps (G.723r53 and G.723r63, respectively). The higher bit rate has greater quality. The lower bit rate provides fair quality and provides system designers with additional flexibility. G.723 CODECs require moderate computation complexity and introduces a relatively high delay.
- **G.729** CODEC samples the filtered voice band at 8000 Hz with a 16 bit resolution, adding compressing algorithm to deliver a stream of 8 Kbps. This CODEC, optimizes the bandwidth used per connection. G.729 CODECs require high computation complexity and introduces a relatively low delay.

The information is transmitted using Real Time Protocol (RTP) over User Datagram Protocol (UDP) over Internet Protocol (IP). The overhead introduced in VoIP communication links by the RTP/UDP/IP headers is quite high: Consider a scenario where a G.729 CODEC operating at a rate of 8Kbps, sending frames every 20 msec. This will result in a voice payloads of 20 bytes for each packet. However, to transfer these voice payloads using RTP/UDP/IP, the following must be added: an Ethernet header of 14 bytes (18 bytes if VLAN is used), IP header of 20 bytes, UDP header of 8 bytes and an additional 12 bytes for RTP. This is a whopping total of 54 bytes (58 with VLAN) overhead to transmit a 20-byte payload. Signaling, which may be based on H.323, SIP, MGCP or MEGACP (H.248), introduces very low bandwidth needs and thus has negligible implications on overall traffic capacity and performances.

Principal VoIP parameters include:

- **Voice Payload Size** (Sample Size): The size in bytes of a packet generated by the CODEC.
- **Packet Size**: The size in bytes of a packet, including RTP/UDP/IP overhead.
- **Packets Per Second** (PPS): The number of packets generated per second. PPS=CODEC Bit Rate/Voice Payload Size.
- **Packet Duration** (Inter Arrival Time or Sample Interval): The time between the start bits of two consecutive packets. Packet Duration=1/PPS.

Most CODECS can be configured to operate at configurable packet duration. The following table summarizes common CODEC characteristics for the smallest packet duration, referred to as basic rate, supported by each CODEC:

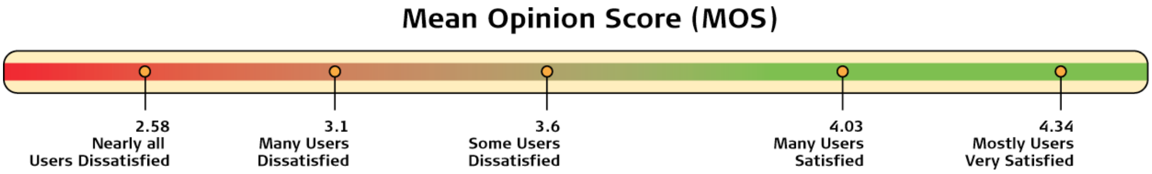
CODEC	Performances at Basic Rate
G.711 A-law or µ-law	200 PPS (5 ms packet duration) 98 bytes packet size (40 Bytes payload + 58 Bytes overhead) Required bandwidth: 156.8kbps per direction
G.729	100 PPS (10 ms packet duration) 68 bytes packet size (10 Bytes payload + 58 Bytes overhead) Required bandwidth: 54.4kbps per direction
G.723r63	33 PPS (30 ms packet duration) 82 bytes packet size (24 Bytes payload + 58 Bytes overhead) Required bandwidth: 21.648kbps per direction
G723r53	33 PPS (30 ms packet duration) 78 bytes packet size (20 Bytes payload + 58 Bytes overhead) Required bandwidth: 20.592kbps per direction

Table 1: common VoIP CODECs' characteristics

MOS (Mean Opinion Score)

In voice communications, particularly Internet telephony, the mean opinion score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance.

To determine MOS, a number of listeners rate the quality of test sentences read aloud over the communications circuit by male and female speakers. A listener gives each sentence a rating as follows: (1) bad; (2) poor; (3) fair; (4) good; (5) excellent. The MOS is the arithmetic mean of all the individual scores, and can range from 1 (worst) to 5 (best). Below is a scale of MOS values and their subjective measured quality.



A MOS of 4.0 is typically the design target. However, in PSTN it may sometime get down to 3.7, and in cellular networks it may get as low as 3.2.

2 When VoIP goes Wireless

Wireless networks have their own advantages and limitations, some of which affect VoIP service performance. When traveling over the wireless IP network, voice packets must contend with phenomena that may affect the overall voice quality. The premier factors that determine voice quality are choice of CODEC, packet loss, latency and jitter.

- **Packet Loss: P**acket loss occurs when packets sent are not properly received by the other end, causing them to be discarded by the receiver. Packet loss can be caused by many different reasons: overloaded links, overload in the receiving device, excessive collisions in the wireless link, physical media errors due to interference or low link quality, and others. Packet loss events may degrade voice quality. Audio CODECs take into account the possibility of packet loss, and most CODECs perform one of several functions that make an occasional packet loss unnoticeable to the user. For example, a CODEC may choose to use the packet received just before the lost packet instead of the lost one, or perform more sophisticated interpolation to eliminate any clicks or interruptions in the audio stream. However, packet loss starts to be a real problem when the percentage of the lost packets exceeds a certain threshold (roughly 5% of the packets), or when packet losses are grouped together in large packet bursts. In those situations, even the best CODECs will be unable to hide the packet loss from the user, resulting in degraded voice quality. Packet loss is more common in wireless than in wire line networks. Typically, mechanisms as retransmissions (ARQ) are used to minimize this phenomenon at the expense of increasing latency and jitter.

- **Jitter:** Typical voice sources generate voice packets at a constant rate. The matching voice decompression algorithm also expects incoming voice packets to arrive at a constant rate. However, the packet-by-packet delay inflicted by the network, caused by reasons such network congestion, timing drift, or route changes, may be different for each packet. Since the receiving decompression algorithm requires fixed spacing between the packets, the typical solution is to implement a jitter buffer within the gateway to compensate for the fluctuating network conditions. The jitter buffer deliberately delays incoming packets in order to present them to the decompression algorithm at fixed spacing. The jitter buffer will also fix any out-of-order errors by looking at the sequence number in the RTP frames. This has the effect of smoothing the packet flow, increasing the resiliency of the CODEC to packet loss, delayed packets and other transmission effects. However, the downside of the jitter buffer is that it can add a significant delay.
- **Latency**: In VoIP links, latency is the mouth-to-ear overall delay. A two-way phone conversation is quite sensitive to latency. Most callers notice round-trip delays when they exceed 250 ms, so the one-way latency budget should typically be lower. The most important components of this latency are:
  - CODEC Latency: Each compression algorithm has certain built-in delay. For example, G.723 adds a fixed 30 ms delay. Choosing different CODECs may reduce the latency, but reduces quality or results in more bandwidth being used.
  - Network Latency: This is the delay incurred when traversing the wireless IP system and the VoIP backbone. It should be possible to specify a higher priority for voice traffic than for delay-insensitive data, in both the wireless and wired parts of the network.
  - Jitter Buffer Size (see above)

When implementing VoIP services over 802.11 based networks, collisions in the wireless link will reduce capacity and increase latency and jitter. This must be taken into account when planning the number of subscribers per sector.

The 802.11 standard has a built-in algorithm that reduces collisions in the wireless medium and thus reduces packet loss. Along with the obvious advantages of this algorithm, some overhead per packet is created. Therefore, proper planning should include traffic planning that reduces the number of packets per second (PPS) generated, thus reducing the overhead affect (further information is provide in following sections).

Question: can we get good VoIP quality over a wireless network?

Answer: Yes! By proper planning and choosing the right VoIP equipment.

3 Choosing the Right VoIP Gateway CPE

In addition to considerations such as preferred signaling method: H.323, SIP, MGCP and MEGACO (H.248) and the required number of voice ports per user site, selecting the right VoIP GW will affect overall solution performances as cell capacity, latency and jitter.

3.1 Cell Capacity

There are two important parameters of VoIP Gateways that affect the number of calls that can be carried by a single sector of a wireless access network:

3.1.1 CODEC (Voice compression)

As wireless access networks provide limited bandwidth per subscriber as compared to a wired LAN for example, compressing CODECs are strongly recommended for saving bandwidth and increasing cell capacity. Using the basic non-compressing G.711 CODEC, which most VoIP gateways support, requires a bandwidth of 156 Kbps per direction at the basic rate (64 kbps raw data per call). In comparison, using compressed G.729 CODEC that delivers a similar quality requires only 54.4 Kbps per direction at basic payload size (8 Kbps raw data), and therefore G.729 is significantly more efficient even at basic payload size.

3.1.2 Packet Duration

Packet duration (Sample Interval) is a very critical consideration when choosing VoIP equipment for a wireless network due to the limitation that the wireless medium imposes on the link in terms of packets per second. (Note that this may happen in wired as well due to the access router capability).

Table 1 specifies, among other things, the number of PPS (packets per second) needed for transmitting voice in each direction at basic rate of each CODEC. When operating with larger payloads, the PPS is reduced. The reduced overhead of a lower PPS results in decreased bandwidth requirement, as can be seen in the following table, but the latency is increased. Therefore, the ability to process more PPS affects directly the required packet duration, system latency and the number of calls that can be supported.

Packet Duration	5 ms	20 ms	30ms	40 ms
PPS per Call (both directions)	400 PPS	100 PPS	66.7 PPS	50 PPS
Packet Size	98 bytes	218 bytes	298 bytes	378 bytes
Required Bandwidth	313.6 Kbps (156.8 Kbps per direction)	174.4 Kbps (87.2 Kbps per direction)	158.9 Kbps (79.5 Kbps per direction)	151.2 Kbps (75.6 Kbps per direction)

Table 2: G.711 CODEC Performances

Packet Duration	10 ms	20 ms	30ms	40 ms
PPS per Call	200 PPS	100 PPS	66.7 PPS	50 PPS
Packet Size	68 bytes	78 bytes	88 bytes	98 bytes
Required Bandwidth	108.8 Kbps (54.4 Kbps per direction)	62.4 Kbps (31.2 Kbps per direction)	46.9 Kbps (23.4 Kbps per direction)	39.2 Kbps (19.6 Kbps per direction)

Table 3: G.729 CODEC Performances

3.2 Latency and jitter

For good voice quality, even under extreme conditions where jitter may be higher than "normal", the VoIP equipment needs a large dynamic jitter buffer. The size of the buffer determines how much jitter (latency variation) can be compensated without degrading the voice quality. The dynamic sizing of that buffer allows latency to be minimized according to the link quality and expected jitter, which may vary also according to network load.

By selecting VoIP gateways with a large and dynamic jitter buffer, one can greatly improve the quality of voice over a wireless network, even without any specific configuration change in the wireless network itself.

4 VoIP over BreezeACCESS VL

4.1 Introduction

BreezeACCESS VL is a high capacity, IP services oriented Broadband Wireless Access (BWA) solution. BreezeACCESS VL employs wireless packet switched data technology to support high-speed WiMAX class services including fast Internet access, Virtual Private Networks, VoIP, gaming, wireless leased lines, hot spot feeding, traffic control, surveillance cameras, telemetric and more. BreezeACCESS VL users are provided with a network connection that is always on, supporting immediate access to the Internet and other IP services at high data rates with high reliability.

The system is designed for cellular-like deployment, enabling the architecture to vary in size and structure. A system can include any number of cells, each containing several Access Units for better coverage of densely populated areas. A Micro Base Station (Stand-Alone Access Unit) is also available for cost-effective support of less populated areas.

The system supports Virtual LANs based on IEEE 802.1Q and 802.3ad Q-in-Q, enabling secure operation and Virtual Private Network (VPN) services and enabling tele-workers or remote offices to conveniently access their enterprise network.

BreezeACCESS VL operates in Time Division Duplex (TDD) mode, using Orthogonal Frequency Division Multiplexing (OFDM) modulation with Forward Error Correction (FEC) coding. Using the enhanced multi-path resistance capabilities of OFDM modem technology, BreezeACCESS VL enables operation in near and non-line-of-sight (NLOS) environments. Additional features such as Adaptive Modulation, use of ARQ, Automatic Power Control, Best AU Selection, Automatic Clear Channel Selection, Automatic Bandwidth Search and different redundancy options are enhancing the robustness of the system. These qualities enable service providers to reach a previously inaccessible and broader segment of the subscriber population and be able to offer reliable video, voice and data services.

BreezeACCESS VL is designed to enable construction of “mixed” cells where it can be used together with other BreezeACCESS products, including BreezeACCESS 900, BreezeACCESS 4900, BreezeACCESS II, BreezeACCESS XL and BreezeACCESS V.

BreezeACCESS VL is currently available in the following frequency bands:

- 4.9 GHz Band: 4.900 – 5.100 GHz
- 5.2 GHz Band: 5.150 – 5.350 GHz

- 5.3 GHz Band: 5.250 – 5.350 GHz
- 5.4 GHz Band: 5.470 – 5.725 GHz
- 5.8 GHz Band: 5.725 – 5.850 GHz

BreezeACCESS VL is the ideal solution for network access as well as backhauling for operators and service provider that want to offer IP-based services.

To better support real time, delay sensitive applications such as VoIP, the system supports a multitude of traffic prioritization mechanisms:

- Layer-2 traffic prioritization based on IEEE 802.1p
- Layer-3 traffic prioritization based on either IP ToS Precedence (RFC791) or DSCP (RFC2474)
- Layer-4 traffic prioritization based on UDP and/or TCP port ranges.

In addition BreezeACCESS VL provides Multimedia Applications Prioritization (MAP) through the Wireless Link Prioritization (WLP) feature: Providing priority in the access to the air link resource, to CPEs with high priority traffic such as VoIP, vs. CPEs with low priority data traffic.

Another important functionality, that helps controlling allocation of network resources based on the operator's business model, is an admission control and dynamic resource allocation mechanism built into the BreezeACCESS-VL: in order to ensure there is no congestion on the VoIP allocated wireless link resources, a special Dynamic Resource Allocation Protocol (DRAP) mechanism that is supported by Alvarion's Voice Gateways and the Access Unit is used. This mechanism ensures that a telephone call will not take place (the user will get a busy tone before dialing attempt), if there are not enough BW resources in the specific sector. This ensures that even in pick hours, service quality is maintained in two levels:

1. MOS is maintained at toll quality (4.0 and above)
2. Data SLAs are not degraded by excess usage of the VoIP service in the same sector.

This mechanism also provides immediate feedback to the end user, in cases where a call cannot be placed. For example, if a sector is set support up to 60 concurrent calls, and a user is trying to establish an additional call, he will get a busy tone as soon as he picks up the handset, indicating that the call cannot be placed at this moment. Without such a mechanism, an excessive number of calls in the sector will cause congestion in the sector resource, high packet loss and degraded call quality to all currently ongoing calls.

Other features that enhance the system capability to support VoIP and make it the ideal solution for applications where high bandwidth, short delay and low error rate are required include:

- System processing power of over 40,000 PPS allows a large number of calls per sector while maintaining enough bandwidth for data services. See section 4.3 for more details.
- Committed Information Rate (CIR) and Maximum Information Rate (MIR) per subscriber unit, per direction.
- Concatenation mechanism enables bundling several frames into a single frame for transmission to the wireless link, improving throughput by reducing the overhead associated with each transmission.

The combination of all these features provides a toll quality telephone service, comparable to landline telephony service. The above capabilities allow the operator to design his optimized network with highly populated sectors (if desired), controlling the amount of resources guaranteed to voice and guaranteed to data SLAs, without sacrificing call quality or meeting SLAs, even in the busiest hours.

4.2 Optimization of BreezeACCESS VL Configuration for VoIP

The goal in this optimization is to:

- Reduce jitter and latency of voice packets
- Reduce the loss of voice packets in case of over-subscription in the sector

The above will directly affect the voice quality of calls in the sector.

4.2.1 Wireless Link Prioritization

The Wireless Link Prioritization feature enables configuring parameters that affect the processes of gaining access to the wireless media and of transmitting high/low priority packets. To optimally support VoIP calls, the Wireless Link Prioritization Option must be enabled. It is also recommended to configure Wireless Link Prioritization parameters and other relevant parameters to support proper prioritization of VoIP traffic vs. data traffic in few scenarios, as follows:

Parameter	10 SUs per sector	50 SUs or more per sector
AU Burst Duration for High Priority Traffic	16 (4 ms)	16 (4 ms)
AU Burst Duration for Low Priority Traffic	20 (5 ms)	10 (2.5 ms)
SU Burst Duration for High Priority Traffic	8 (2 ms)	8 (2 ms)
SU Burst Duration for Low Priority Traffic	20 (5 ms)	10 (2.5 ms)
Number of HW Retransmissions for Low Priority Traffic	10	10
Number of HW Retransmissions for Low Priority Traffic	10	10
Low Priority AIFS	3	3
Concatenation	Enabled	Enabled
Maximum Concatenated Frame Size	4032 bytes	4032 bytes
Minimum Contention Window	15	255
Maximum Contention Window	1023	1023

Table 4: Recommended Configuration of BreezeACCESS VL Parameters

4.2.2 QoS through Priority Queuing

The BreezeACCESS VL supports low, mid and high priority queuing. Mid queue is used only for management and broadcast/multicast traffic. Mission critical traffic, such as VoIP, should be mapped to the highest priority queue, and best effort traffic should be mapped to the low priority queue. This ensures that even if there are many long data packets waiting their turn to be transmitted over the air, the high priority voice packet will be transmitted first. Moreover, this can reduce packet loss of VoIP packets in conditions where the network is congested with low priority data packets.

To have prioritization properly functioning, it is required to use the same prioritization method in the VoIP Gateway and in BreezeACCESS VL. Priority tagging in the Gateway should be in accordance with the high and low priority threshold configured in the BreezeACCESS VL.

4.2.3 Admission Control

To enjoy the benefits of the Dynamic Resource Allocation Protocol implemented in BreezeACCESS VL and Alvarion's Gateways, the DRAP Option must be enabled. The Number of Voice Calls parameters that set the upper limit for calls allowed in the sector should be set according to the number of voice ports in the cell and the operator's service policy. For details on the number of voice calls that can be supported in different scenarios without sacrificing quality is provided in section 4.3. When deciding on the maximum number one should take into account the average modulation level in the sector.

4.2.4 Bandwidth Management

An important feature of BreezeACCESS VL, which can help improve the voice quality and capacity, is its ability to control the amount of traffic that each station can forward to the wireless link. The target is to limit the over-subscription of a BreezeACCESS VL sector, to the lowest possible value.

For VoIP applications the recommended bandwidth settings are:

1. Taking the sum of the MIR settings of all SUs in the sector, the total MIR should be as close as possible to the maximum bandwidth available in the sector (the actual bandwidth which is the mean of the modulation level of all SUs in that sector), with as little over-subscription as possible.
2. Allocate a CIR value to each SU that is at least equal to the bandwidth required per voice call multiplied by the number of calls expected from this SU. This will ensure that if the sector is congested, the SU will still receive air bandwidth allocation for voice calls.

4.2.5 Sector Load

As stated above, the number of active SUs in a BreezeACCESS VL sector will affect its capacity as well as the latency and jitter. Although latency and jitter are of minor importance to data services, they do have significant relevancy for VoIP applications.

Sector capacity, defined as the transmit/receive throughput an access unit can provide, depends on average modulation used for sending and receiving the packets (each CPE can have different modulation per direction depending on its link quality) and interferences (affect retransmissions). Therefore, the relevant parameters when planning for over-subscription in a VoIP enabled sector should include:

1. Capacity over-subscription (total MIR and CIR allocated to all SUs in the sector vs. available capacity).
2. Number of concurrently active data sessions and characteristics of these sessions.
3. Estimated percentage of concurrently active voice calls (vs. the total number of POTS ports, using standard tools such as Erlang B).

Another parameter, which can increase sector load without degrading voice quality, is voice packet payload: the higher the payload, the lower the required bandwidth and processing power.

4.3 BreezeACCESS VL Calls Capacity

BreezeACCESS VL presents a break through in performances and future ready capacity, with up to 288 calls and up to 40Mbps UDP data rate, per sector. With BreezeACCESS VL operators and ISPs can offer time critical and high throughput services.

Table 7 lists the maximum achievable calls capacity of a BreezeACCESS VL sector based on experience and simulations. These performances are for an "ideal" sector where Rate (Modulation Level) 8 can be used by all CPEs.

BW [MHz]	Rate	Codec	VoIP Packing Duration	CPEs	Calls per CPE	Concurrent Calls	MOS	Delay introduced by VL	Mouth to Ear Delau [msec]	Remaining FTP Data Capacity
Residential										
20	8	G.711	20 ms	50	1	41	4.3	10	90	no data
		G.729	30 ms	50	1	50	4.02	2	122	no data
10	8	G.711	20 ms	50	1	20	4.3	8	88	no data
		G.729	30 ms	50	1	32	4.02	7	127	no data
Business										
20	8	G.711	20 ms	7	24	168	4.17	28	108	no data
		G.729	30 ms	12	24	288	4.01	14	134	no data
		G729	30 ms	4	8	32	4	6	126	6 Mbps/CPE
		G729	30 ms	6	2	12	4.02	6	126	4 Mbps/CPE
		G729	30 ms	8	2	16	4.02	6	126	3 Mbps/CPE
10	8	G.711	20 ms	1	24	24	4.38	6	86	no data
		G.729	30 ms	5	24	120	4	18	138	no data

Table 5: Maximum Achievable Call Capacity in best conditions

In realistic scenarios, the required call capacity is by far lower than the numbers presented in Table 5. A significantly higher number of CPEs can be supported without compromising call quality. It should be noted that translation of the number of concurrent calls to actual number of registered users depends on operator's over-subscription policy. Table 6 below provides the number of VoIP CPEs that can be supported when using for example Erlang B for the voice service model.

Number of Calls	Erlang B, assuming 0.01 blocking probability	Residential Users, Assuming 0.07 Erlang (4.2 minute call in average)	Business Users Assuming 0.15 Erlang (9 minute call in average)
2	0.15	2	1
4	0.87	12	6
6	1.91	27	13
8	3.13	45	21
10	4.46	64	30
12	5.88	83	39
15	8.11	116	54
20	12.03	172	80
25	16.13	230	108
30	20.34	291	136
35	24.64	352	164
40	29.01	414	193
45	33.43	478	223
50	37.90	541	253

Table 6: Registered VoIP users based on Erlang B model

The following tables display concurrent calls capacity of a typical good-quality BreezeACCESS VL sector, with Rate 5 as the average Modulation Level, serving residential users (Table 7) or business customers (Table 8).

Note that these tables assume a target MOS of 4.0 or higher, representing a quality that satisfies most users. If the requirement for quality is reduced, a higher number of simultaneous calls can be supported.

BW [MHz]	Rate	Codec	VoIP Packing Duration	CPEs	Calls per CPE	Concurrent Calls	MOS	Delay introduced by VL	Mouth to Ear Delay [msec]	Remaining FTP Data Capacity
20	5	G.711	20 ms	50	1	10	4.37	5	85	8.1
					1	15	4.33	5	85	5.5
		G.729	30 ms	50	1	10	4.02	4	124	10.3
					1	15	4.02	5	125	8
10	5	G.711	20 ms	25	1	6	4.36	6	86	3.5
					1	10	4.36	6	86	1.6
				50	1	10	4.34	7	87	2.3
					1	15	4.33	8	88	0.5
		G.729	30 ms	25	1	6	4.02	5	125	4.7
					1	10	4.02	6	126	3.1
				50	1	10	4.02	6	126	3.5
					1	15	4.02	6	126	2

Table 7: Call Capacity, Residential Users

BW [MHz]	Rate	Codec	VoIP Packing Duration	CPEs	Calls per CPE	Concurrent Calls	MOS	Delay introduced by VL	Mouth to Ear Delay [msec]	Remaining FTP Data Capacity per CPE [Mbps]
20	5	G.711	20 ms	10	2	8	4.37	5	85	12.4
					8	20	4.37	2	82	7.2
					24	50	4.24	9	89	0.7
				20	2	15	4.38	3	83	3.9
					8	35	4.34	6	86	0.5
		G.729	30 ms	10	2	8	4.01	6	126	13.7
					8	20	4.02	2	122	10.6
					24	50	4	6	126	5.1
				20	2	15	4.02	2	122	5.9
					8	35	4.02	3	123	2.7
					24	88	4.01	10	130	0.2
10	5	G.711	20 ms	10	2	8	4.37	6	86	4.8
					8	20	4.32	7	87	0.5
				20	2	15	4.35	6	86	0.5
		G.729	30 ms	10	2	8	4.02	6	126	6.2
					8	20	4.02	4	124	3.4
					24	50	4	12	132	0.3
				20	2	15	4.02	4	124	2
					8	35	3.73	28	148	0

Table 8: Call Capacity, Business Users

4.4 Examples

4.4.1 Example I

Requirements

- A package service for residential users with one voice port and best effort data
- Data service required is 4096 kbps downstream and 256 kbps upstream. Maximum over subscription of 1:25
- VoIP service with MOS > 4.0

Expecting service penetration of 150 residential subscribers per base station

Calculations:

- Using 20 MHz channels
- Base stations is constructed with 3 sectors each, hence ~50 subscribers per sector
- For VoIP, selecting G.729 with 30 ms packet duration. Based on Table 6, for 50 users with one voice port it is required to support 10 concurrent calls
- Assuming an average sector rate of 5 (a very modest assumption for planning purposes, expecting actual sector rate to be higher in reality) and using Table 7, residual sector data capacity after securing bandwidth for 10 calls is 10.3 Mbps
- 4096 Kbps plus 256 Kbps, times 50 users, divided by 25 because of over subscription, results in BW requirements of 8.7 Mbps, which is below the available 10.3 Mbps. Hence, data capacity will be sufficient to support the requirements.
- CPE downstream Maximum Information Rate (MIR) is to be set to 4096 Kbps
- CPE upstream MIR is to be set to 256 Kbps
- CPE downstream Committed Information Rate (CIR) is to be set to zero
- CPE upstream CIR is to be set to zero

Conclusions: With this scenario, 150 residential users per base station can be supported with margins, taking into account the capacity and quality required for the service.

4.4.2 Example II

Requirements:

- A package service for businesses with eight voice ports and data service
- Data service required is 2048 Kbps bi-directional with over subscription of 1:4. It is also required that CIR will be 512 Kbps per direction

- VoIP service with MOS > 4.0

Expecting service penetration of 30 businesses per base station

Ca**lculations**:

- Using 20 MHz channels
- Base stations is constructed with 3 sectors each, hence ~10 business subscribers per sector
- For VoIP, selecting G.729 with 30 ms packet duration. Based on Table 6, for 80 Voice ports (10 businesses with 8 calls each) it is required to support 20 concurrent calls
- Assuming an average sector rate of 5 (a very modest assumption used for planning, expecting actual sector rate to be higher in reality) and using Table 8, residual sector data capacity after securing bandwidth for 20 calls is 10.6 Mbps
- 2048 Kbps bi-directional, times 10 businesses, divided by 4 because of over subscription, results in BW requirements of 10.24 Mbps, which is below the available 10.6Mbps. Hence, data capacity will be sufficient to support the requirements.
- CPE MIR per direction is to be set to 2048 Kbps
- CPE CIR per direction is to be set to 512 Kbps (512 Kbps bi-directional times 10 businesses is 10.24 Mbps which is below 10.6 Mbps, hence no issue to support this CIR requirement)

Conclusions: With this scenario, 30 enterprises per base station can be supported with margins, taking into account the capacity and quality required for the service.

5 Summary

VoIP telephony is an excellent technology that allows operators to offer telephony services over their existing wireless networks.

As shown in this document, the right VoIP equipment that can support all the right functionality can have a critical impact on the capacity and performance of the telephony application over the wireless network. By optimizing both the VoIP equipment and the wireless infrastructure, voice capacity can be increased significantly. Making a difference that can absolutely improve the business model and turn the application to a very worthwhile revenue generator. Moreover, by correctly configuring and building the wireless IP network, good telephony quality can be achieved, and overall voice and data capacity can be increased.