# White Paper

# Improving the Quality of Communications with Packet Loss Concealment

In digital transmission of speech and audio signals, lost or corrupted frames and packets can severely degrade audio quality at the receiver. A special technique known as Packet Loss Concealment (PLC) is used to compensate for these quality-robbing effects. This white paper introduces the PLC technologies developed by Broadcom for applications such as Voice over Internet Protocol (VoIP), DECT cordless phones, and Bluetooth® communication. These advanced techniques provide dramatic improvements to audio quality under packet-loss conditions.

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# Introduction

In digital transmission of speech or audio signals, a coder encodes a speech or audio signal into a digital bit stream for transmission. A decoder decodes the digital bit stream into a speech or audio signal. The combination of the *co*der and the *dec*oder is called a *codec*. The transmitted bit stream is usually organized into segments called frames. In packet transmission networks, each transmitted packet may contain one or more frames of a compressed bit stream.

In wireless or packet networks, sometimes the transmitted frames or packets are erased or lost. This condition is called *frame erasure* in wireless networks and *packet loss* in packet networks. When this condition occurs during the active portion of the audio signal, audio quality is severely degraded. To preserve audio quality, the decoder needs to perform a special algorithm to try to conceal the degradation caused by the lost frames. This kind of algorithm is called *frame erasure concealment* (FEC) or *packet loss concealment* (PLC), depending on the communication system in which it is being used. Since the terms FEC and PLC generally refer to the same kind of technique, PLC will be used to refer to both in this white paper.

The basic ideas of PLC date back to at least early 1980s[1]. In the last 15 years there has been a substantial increase in research of PLC techniques, presumably because voice and audio signals are increasingly being transmitted as packets. While early PLC research in the 1980s dealt mostly with memoryless codecs that are based on Pulse Code Modulation (PCM), most of the recent research on PLC since the early 1990s has been concentrated on predictive speech codecs and frequency-domain audio codecs, such as transform codecs and sub-band codecs.

Some of the pioneers of the earliest PLC techniques for predictive speech codecs[2] have been working for Broadcom since 2000, and in the last seven years several more experts in voice and audio processing technologies have been recruited. Because different applications have different sets of requirements, no single PLC algorithm can be the optimal solution for all applications. Instead, different PLC algorithms need to be created to accommodate the different codec and application requirements. With over 70 years of combined experience in related fields, these engineers have developed six families of PLC algorithms optimized for different applications:

- BroadVoice®16/32 PLC
- BroadVoice Concealment (BVC)
- Audio-BVC
- ITU-T G.722 Appendix III
- Reduced-Complexity BVC (RC-BVC)
- Low-Complexity PLC (LC-PLC) in SmartAudio™

Most of these PLC algorithms have been successfully deployed in many commercial Broadcom chips, such as Bluetooth and VoIP chips. This includes a G.722 PLC algorithm that was rated significantly better than all other candidates in ten official ITU-T subjective listening tests conducted in three countries. The superior performance of the Broadcom PLC algorithm led the ITU-T to adopt it as an industry standard (Recommendation G.722 Appendix III) in 2006, confirming Broadcom’s leadership in PLC technologies.

1. See references [1], [2], and [3].
2. See references [4] and [5].

This white paper aims to give a general introduction to PLC in general, and to these six families of PLC algorithms in particular, so that readers can have an understanding and appreciation of Broadcom's dominant PLC technologies.

# Overview of Packet Loss Concealment

Audio signals are transmitted as packets in many VoIP and Bluetooth applications. A packet is considered lost when it is literally lost, is corrupted to the point that it cannot be used, or arrives too late to be useful. When a packet is lost, a large block of the digital bit stream is missing. What should the decoder do in this case? The simplest method is to fill in zeros as the output signal samples in the frames corresponding to the lost packets. This "zero-fill" method creates discontinuities and gaps in the output audio waveform and severely degrades the output sound quality.

A slightly better approach is to ramp down the magnitude of the output audio signal toward zero at the beginning of a segment of lost packets and then ramp up the signal magnitude at the end of the segment. This approach was commonly used for voice communication between a Bluetooth headset and cell phone handset. Unfortunately, the output audio quality of this "ramp-down-to-zero" approach is only marginally better than the "zero-fill" approach, and the resulting audio quality is still very poor.

"Frame Repeat" is another simplistic approach in which the last received "good" frame of a compressed bit-stream is repeated to replace the missing packet, and the repeated frame is then decoded as if it were the correctly received frame. The effectiveness of this approach is highly dependent on the codec that is used and the characteristics of the audio signal when the packet loss occurs. However, in general the frame-repeat approach falls short of seamless concealment of packet loss, and the output audio signal will contain audible artifacts that diminish the user experience.

The three simplistic methods referenced above are often considered not to be true PLC techniques. A true PLC algorithm generally uses more sophisticated signal processing techniques to achieve much better output audio quality than is possible with the methods above. The goal is for the listener to be mostly unaware of packet loss, or at least to make the audio distortion as small as possible. This is often achieved by extrapolating previously decoded signals to fill in the waveform gap corresponding to the packet loss, and by taking special care to ensure a smooth waveform transition at the boundaries between received frames (good frames) and missing frames (bad frames).

Figure 1 shows the speech waveforms of a female utterance of "*On the...*" for decoded speech without packet loss and the output waveforms of the four methods discussed above with 10% random packet loss. The figure shows that the zero-fill and ramp-down-to-zero methods introduce distinct waveform gaps wherever packet loss has occurred. The figure also shows that the frame-repeat method often fills in the waveform gaps with replacement waveforms that are out of phase with the actual missing waveforms. The output speech obtained by using the three simpler methods sounds rough and highly distorted. In contrast, the output of the waveform-extrapolation-based PLC looks almost the same as the decoded speech with no lost packets; it even sounds essentially the same and is free of any noticeable distortion, at least for this particular speech segment.

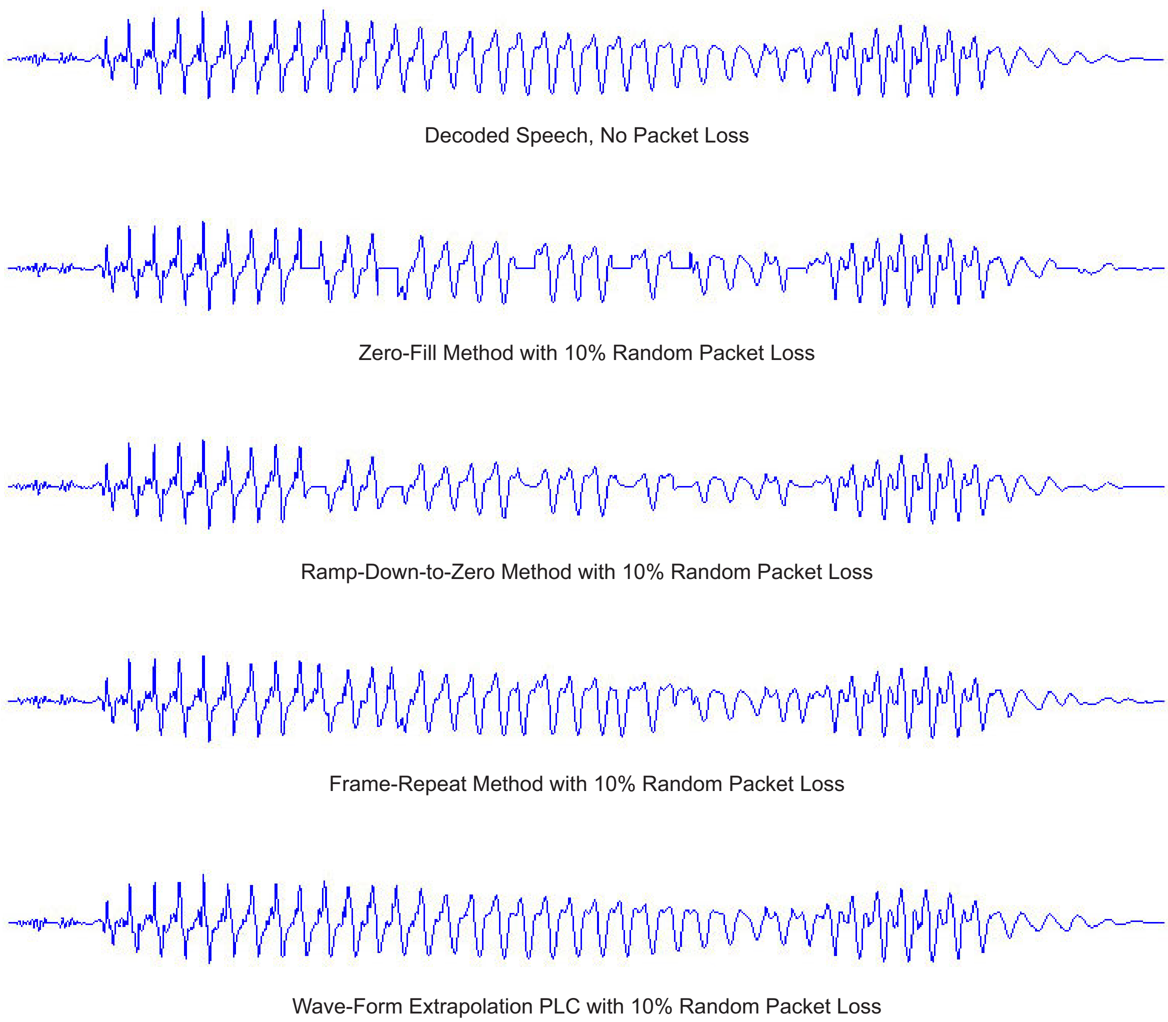


**Figure 1: Speech Waveform Comparison Between Different Packet-Loss Handling Methods**

Extrapolation methods are typically used for PLC, because voice communication systems need to minimize the delay (latency) of the audio signal and normally do not allow PLC to add any delay. In some applications where additional delay is allowed and PLC can wait for the arrival of the next good frame, it is possible to use interpolation to achieve better audio quality. Nevertheless, most published PLC research papers have been based on extrapolation methods because they assume that little or no additional delay is allowed.

PLC normally does not require any change in the operation of the encoder because it is usually applied as a post-processing step after the decoding operation. Consequently, different vendors can apply different proprietary PLC techniques to a given codec without affecting bit-stream compatibility.

Most PLC techniques are tightly integrated with the decoder for two reasons: first, to reduce complexity by re-using the intermediate results calculated by the decoder, and second to help the decoder recover from packet loss as quickly as possible. The constraints imposed by this tight integration usually demand that a specific PLC algorithm be developed for each speech or audio decoder, although there are exceptions. A PLC algorithm developed for a specific codec can also be applied to another codec with no changes or only minor changes if the decoder used in the other codec is sufficiently similar to the one used in the codec for which the algorithm was originally designed.

In the literature, many PLC techniques have been proposed, and many were adopted by standardizing bodies. For example, the ITU-T Recommendation G.711 Appendix I specifies a PLC algorithm for the G.711 PCM codec[3]. When packet loss occurs, this algorithm identifies the "pitch" period (in the case of voiced speech segments) or the time period of maximum waveform similarity (in the case of unvoiced or transitional segments) using decoded signals in the immediate past, then periodically repeats the decoded waveform at that pitch period. An additional delay of 3.75 milliseconds is required for an overlap-add operation to smooth out waveform discontinuities that would otherwise occur.

Most of the modern speech codecs standardized in the last 20 years are based on the predictive coding principle, and most of them have PLC specified as part of the standard. These PLC standards usually extrapolate the decoded excitation waveform by periodic or random repetition, repeat or extrapolate gain and filter parameters, and then pass the extrapolated excitation signal through the extrapolated synthesis filters to synthesize the output audio signal. There are also other PLC approaches based on the sinusoidal model of audio signals or the Hidden Markov Model (HMM). However, these other approaches have not gained wide acceptance, perhaps due to their higher complexity.

# Overview of Broadcom's PLC Technologies

A brief overview of the six families of Broadcom's PLC algorithms is given below in the order they were developed.

## BroadVoice16/BroadVoice32 (BV16/BV32) PLC

This is Broadcom's first PLC technology, designed for predictive speech codecs called BroadVoice16 (BV16) and BroadVoice32 (BV32), which are 8 kHz narrowband and 16 kHz wideband standard speech codecs, respectively. These two codecs are specified in PacketCable, SCTE, ANSI, and ITU-T standards. The BV16/BV32 PLC mitigates packet loss better than most other PLC techniques that have been standardized for codecs.

## BroadVoice Concealment (BVC)

This PLC algorithm is derived from BV16/BV32 PLC and is specially optimized for memoryless codecs such as a G.711 PCM. Compared with ITU-T G.711 Appendix I, BVC performs significantly better, uses significantly less RAM memory, and eliminates the 3.75 milliseconds of additional delay.

## Audio-BVC

This PLC algorithm is based on BVC and is optimized for transform audio codecs based on MDCT (Modified Discrete Cosine Transform) or memoryless codecs. Designed to be a "universal" PLC, this Audio-BVC algorithm performs well not only for speech, but also for music and general audio signals,

3. This PLC algorithm is based on the ideas proposed in reference [3].

more so than the original BVC. A variety of techniques were developed to enable Audio-BVC to achieve a relatively low complexity at a high sampling rate of 48 kHz.

### ITU-T G.722 Appendix III

This algorithm is based on BVC, and was submitted to the ITU-T for G.722 PLC competition. Significant improvements were added to BVC algorithm, including some that were designed specifically to handle the peculiar behavior of the backward-adaptive ADPCM decoder inside the G.722 wideband codec after packet loss has occurred. This algorithm achieved much better performance than other candidates in the ITU-T official listening tests, and was selected as the standard for ITU-T G.722 Appendix III.

### Reduced-Complexity BVC (RC-BVC)

Compared with the original BVC, RC-BVC code size was reduced by almost 20%, and computational complexity was reduced nearly 50% while maintaining essentially the same output audio quality.

### Low-Complexity PLC (LC-PLC) in SmartAudio

This PLC algorithm was developed specifically for Bluetooth applications and is part of Broadcom's SmartAudio algorithms for Bluetooth audio enhancement. It was developed from scratch rather than derived from BVC. It achieves a very low code size and low computational complexity while maintaining essentially the same level of audio output quality as BVC.

Each of these six families of PLC algorithms will be discussed in more detail in the subsequent sections.

## BroadVoice16/32 PLC

BroadVoice16 (BV16) is a 16 Kbps[4] speech codec for 8 kHz sampled narrowband speech. BroadVoice32 (BV32) is a 32 Kbps speech codec for 16 kHz sampled wideband speech[5]. Both BV16 and BV32 were developed by Broadcom, and both were standardized for VoIP applications for cable telephony by CableLabs in PacketCable 1.5/2.0, by SCTE (Society of Cable Telecommunications Engineers) and ANSI (American National Standards Institute) in ANSI/SCTE 24-21 2006 and ANSI/SCTE 24-23 2007, and by the ITU-T in Recommendations J.161 and J.361, respectively.

As part of the BV16 and BV32 development effort, Broadcom created proprietary PLC algorithms for BV16 and BV32[6]. At the request of CableLabs, Broadcom also developed "example PLC" for BV16 and BV32 during the standardization of BV16 and BV32 in PacketCable 1.5 and 2.0. Broadcom's proprietary PLC for BV16 and BV32 performs slightly better than the example PLC in the BV16 and BV32 standards. The proprietary PLC algorithms for BV16 and BV32 have been ported to Broadcom's VoIP chips for IP phones, cable voice gateways, and DSL voice gateways.

The BV16 and BV32 encoders are based on Two-Stage Noise Feedback Coding[7]. Although they differ from the encoders used in most other standard predictive speech codecs that are based on Code Excited

4. In this context, Kbps means 1000 bits per second.
5. See reference [6].
6. See reference [7].
7. See reference [8].

Linear Prediction (CELP), the decoders in the BV16 and BV32 are still similar to those used in other standard CELP codecs. As a result, it is possible, with minor modifications, to use the BV16/BV32 PLC algorithm as a replacement for the native PLC algorithms in other CELP codecs.

Unlike conventional PLC for standard predictive speech codecs, which extrapolate the excitation signal, Broadcom's proprietary PLC for BV16 and BV32 extrapolates the speech signal directly and uses an innovative "filter ringing" method and overlap-add to eliminate waveform discontinuities at frame boundaries. The filter ringing method smoothly extends the speech waveform from the last frame to the current frame and eliminates the 3.75 millisecond overlap-add delay associated with G.711 Appendix I. After waveform extrapolation, the proprietary BV16/32 PLC then updates filter memories in bad frames by inverse filtering, and also attempts to correct filter memories at the arrival of the first good frame after packet loss[8].

To keep the complexity low, the proprietary BV16/BV32 PLC makes as much use as possible of what is already available in the BV16/32 decoder. For example, the speech waveform extrapolation used in the proprietary PLC is based on the pitch period already available in the BV16/32 decoder, and filter ringing is calculated using the long-term and short-term predictor parameters already decoded in the BV16/32 decoder. Therefore, only a minimal amount of extra complexity is required by the proprietary PLC. On a ZSP400 DSP core, the worst-case additional complexity is only a small fraction of one MHz for the BV16/ BV32 PLC.

In terms of performance, most narrowband standard speech codecs degrade Mean Opinion Score (MOS) in formal subjective listening tests by 0.5 at around 3% random packet loss rate, but the BV16 PLC withstands almost 5% random packet loss before losing 0.5 point of Mean Opinion Score (MOS)[9]. Similarly, most wideband standard codecs lose 0.5 MOS at 1–2.5% packet loss, but the BV32 PLC withstands 5.3% packet loss before losing 0.5 MOS. In other words, the PLC for BV16 and BV32 can compensate for a higher level of packet loss than most other PLC techniques for standard codecs, since it can withstand a higher packet loss rate before 0.5 MOS degradation is reached.

# BroadVoice Concealment (BVC)

BroadVoice Concealment (BVC) is a PLC algorithm optimized specifically for memoryless codecs such as linear PCM or G.711 logarithmic PCM. Broadcom originally developed BVC as a better-performing replacement for the ITU-T G.711 Appendix I, the standard PLC scheme for G.711 PCM. BVC was created by modifying the PLC for BV16 and BV32 to get the narrowband BVC (NB-BVC) and wideband BVC (WB-BVC), respectively, for 8 kHz and 16 kHz sampling rates. Many of the features in the BV16/BV32 PLC that do not apply to memoryless codecs, such as filter memory update and filter memory correction, are eliminated in BVC.

BVC addresses three issues with G.711 Appendix I: the 3.75 ms additional delay; the large RAM requirement, partially due to the need to buffer 390 samples of speech signal; and insufficient quality of output speech. The 3.75 ms delay is avoided by using the filter ringing approach carried over from the BV16/BV32 PLC. The RAM requirement was greatly reduced in BVC compared with G.711 Appendix I. Finally, the output speech quality has been significantly improved.

8. See reference [7].
9. See reference [9].

Unlike BV16/BV32 PLC, which can use the pitch period and filter parameters already decoded by the BV16/BV32 decoder, BVC is stand-alone and needs to compute these parameters independently. As a result, the computational complexity of BVC is somewhat higher than the incremental complexity of BV16/BV32 PLC. Regardless, the complexity of BVC is still relatively low—only in the low single-digits in terms of MHz.

There are two phases of BVC development. Phase 1 BVC does not require any additional delay and uses only waveform extrapolation from previous frames. Phase 2 BVC allows one additional frame of delay and may use the information in the next frame if it is received as a good frame. In the end, Phase 1 BVC satisfied the needs of customers and has been ported to Broadcom chips, while Phase 2 BVC remains in the research stage and has not been ported to any chips.

The performance of Phase 2 BVC in terms of PESQ (ITU-T P.862 Perceptual Evaluation of Speech Quality) is reported in Figure 2 together with the PESQ scores of the zero-fill method, G.711 Appendix I, and Phase 1 BVC. All PESQ curves in the figure were measured with random packet loss and a packet size of 10 milliseconds. The figure shows that for higher packet loss rates, Phase 1 BVC is about 0.2 PESQ higher than G.711 Appendix I, which represents a significant improvement. Listening comparisons confirmed that Phase 1 BVC sounded noticeably better than G.711 Appendix I. It can also be seen that for higher packet loss rates, Phase 2 BVC provides another 0.2 PESQ improvement over Phase 1 BVC and reaches a total improvement of 0.4 PESQ over G.711 Appendix I—a very large improvement indeed. The PESQ advantage provided by Phase 2 BVC over G.711 Appendix I is almost the same as the improvement provided by G.711 Appendix I over the zero-fill method.

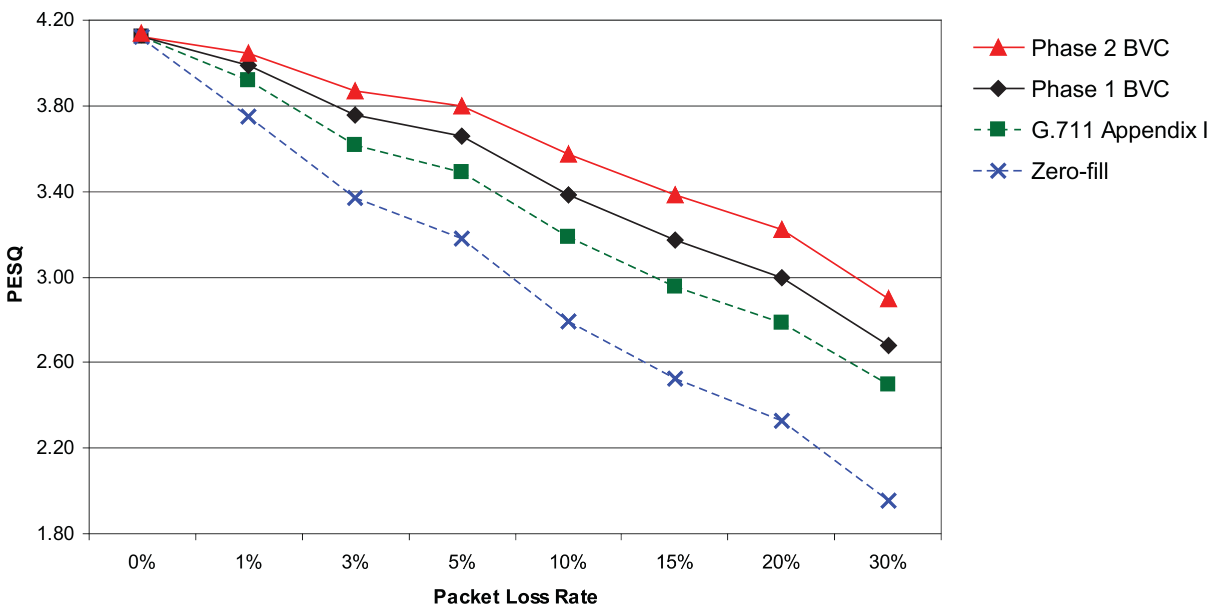


**Figure 2: PESQ comparison of BVC and G.711 Appendix I**

# Audio-BVC

Most PLC methods described in this white paper are optimized for speech codecs and operate at a relatively low sampling rate of 8–16 kHz. For some applications in which music or general audio signals sampled at 44.1 kHz or 48 kHz are transmitted as packets, there is a need for a PLC algorithm that can handle general audio signals and the much higher sampling rates without increasing the complexity to an unsustainable level. The challenges here are two-fold. First, while periodic waveform extrapolation works well for voiced speech signals (for instance, vowel sounds), it doesn't work well for complex music or general audio signals. Second, a major portion of the computational complexity of PLC is in finding a suitable time-lag for waveform extrapolation, and that part of the complexity is normally proportional to the square of the sampling rate. As such, going from 8 kHz to 48 kHz can easily increase that part of the complexity by a factor of thirty-six[10] unless special care is taken to limit the growth of complexity with the higher sampling rate. Broadcom has developed a PLC algorithm called Audio-BVC in a major development effort to address both challenges.

As the name implies, Audio-BVC is a PLC algorithm derived from BVC but is optimized for general audio signals and for audio codecs that may operate at a sampling rate up to 48 kHz or higher. A substantial number of innovations went into the Audio-BVC algorithm to keep the computational complexity relatively low and to re-use BVC code to minimize the increase of the code size. How did the complexity reduction effort pay off? For the MPEG AAC codec at 48 kHz sampling, the corresponding Audio-BVC is estimated to take less than 10 MHz on the ZSP400 DSP core for bad frames, and substantially less for good frames. This represents only about 2.5 times the complexity of narrowband BVC rather than the 36 times mentioned earlier.

Most codecs used for coding general audio signals are adaptive transform codecs based on MDCT, although some are based on sub-band codecs or even PCM. Audio-BVC is designed to work with either MDCT-based audio codecs or memoryless PCM. When it is used with PCM, the filter ringing method of BVC is used to smooth the waveform transition near frame boundaries. MDCT-based audio codecs typically have a built-in overlap-add operation with 50% window overlap. In the event of packet loss, Audio-BVC takes full advantage of this built-in overlap-add operation to get the most out of the information embedded in the bit-stream of the last good frame. The result is a dramatic improvement in performance.

Audio-BVC also employs a variety of PLC techniques to handle different kinds of audio signals, and it uses sophisticated signal analysis and classification to pick the PLC technique that is most suitable for concealing the effects of the packet loss for the current frame of audio signal. Compared with pure BVC, Audio-BVC is more "universal" and performs more robustly across all kinds of audio signals such as pure speech, pure music, pure background noise, speech in background music, speech in background noise, and more.

# ITU-T Recommendation G.722 Appendix III

In early 2006, the European Telecommunications Standards Institute (ETSI) asked the ITU-T to standardize a PLC algorithm for the ITU-T G.722 sub-band ADPCM wideband speech codec. This followed a decision by ETSI to use G.722 as the wideband codec for the next-generation DECT cordless telephones. Broadcom, France Telecom, and OKI Electric Industry of Japan each developed a candidate

10. The formula used to calculate this value is $(48/8)^2$.

G.722 PLC algorithm and submitted it to the ITU-T in August 2006. The Broadcom candidate was rated significantly better than the others in ten official ITU-T listening tests conducted in three countries. The Broadcom candidate was standardized as G.722 Appendix III in November 2006, reconfirming Broadcom's industry leadership in PLC technology.

Broadcom's G.722 Appendix III PLC algorithm[11] uses the wideband BVC algorithm to extrapolate the full-band speech waveform. In addition, it employs a set of novel techniques to "tame" the peculiar behaviors of the backward-adaptive ADPCM codec in each sub-band after a packet loss event. This includes increasing the safety margin for stability, performing smoothing and averaging of internal variables, and placing special constraints on parameter values, among others. Furthermore, it uses a time-warping technique to warp the time axis of the first good decoded frame after packet loss to align it with the extrapolated waveform, and also uses a "re-phasing" technique to adjust the end point of G.722 re-encoding so that the "phase" of the decoder states update aligns with the first decoded good frame after packet loss[12].

During G.722 PLC standardization, the additional complexity of G.722 Appendix III on top of the G.722 decoder has been measured using C source codes with an ITU-T software tool. The computational complexity was measured in terms of Weighted Million Operations Per Second (WMOPS). The complexity numbers have been cross-checked by France Telecom and are shown in Table 1. The G.722 Appendix III PLC algorithm has been ported to Broadcom's VoIP-related chips.

**Table 1: Complexity of G.722 Appendix III**

| | |
|---|---|
| Program size (kwords) | 2.00 |
| Data tables (kwords) | 0.64 |
| Per-instance RAM (kwords) | 1.01 |
| Scratch RAM (kwords) | 0.59 |
| Worst-case WMOPS (10 ms packet) | 2.78 |
| Worst-case WMOPS (20 ms packet) | 2.50 |
| Average WMOPS (10 ms packet) | 1.98 |
| Average WMOPS (20 ms packet) | 1.97 |

In terms of performance, the results of the ITU-T official subjective listening tests indicate that G.722 Appendix III is more robust than the PLC of the G.729.1 wideband codec when comparing the MOS degradation from the clear-channel MOS as a function of the packet loss rate for both random packet loss and burst packet loss. Furthermore, Dynastic Inc. did a global statistical analysis of the results of the ten official ITU-T tests and summarized the test results from those listening tests (Figure 3) for the four experiments in the ITU-T test plan: Experiment 1a, clean speech with random packet loss; Experiment 1b, clean speech with burst packet loss; Experiment 2a, speech in background music; and Experiment 2b, speech in background noise. In all four figures, PLC-A is the Broadcom candidate, PLC-B is the OKI candidate, PLC-C is the France Telecom candidate, and PLC-0 is a reference condition. Dynastat's conclusion from the global statistical analysis of the results from all ten listening tests was that "*PLC-A is clearly the best performing PLC across the subjective experiments involved in the G.722 PLC Selection Test.*"[13].

11. See reference [10].
12. See reference [11].
13. See reference [12].

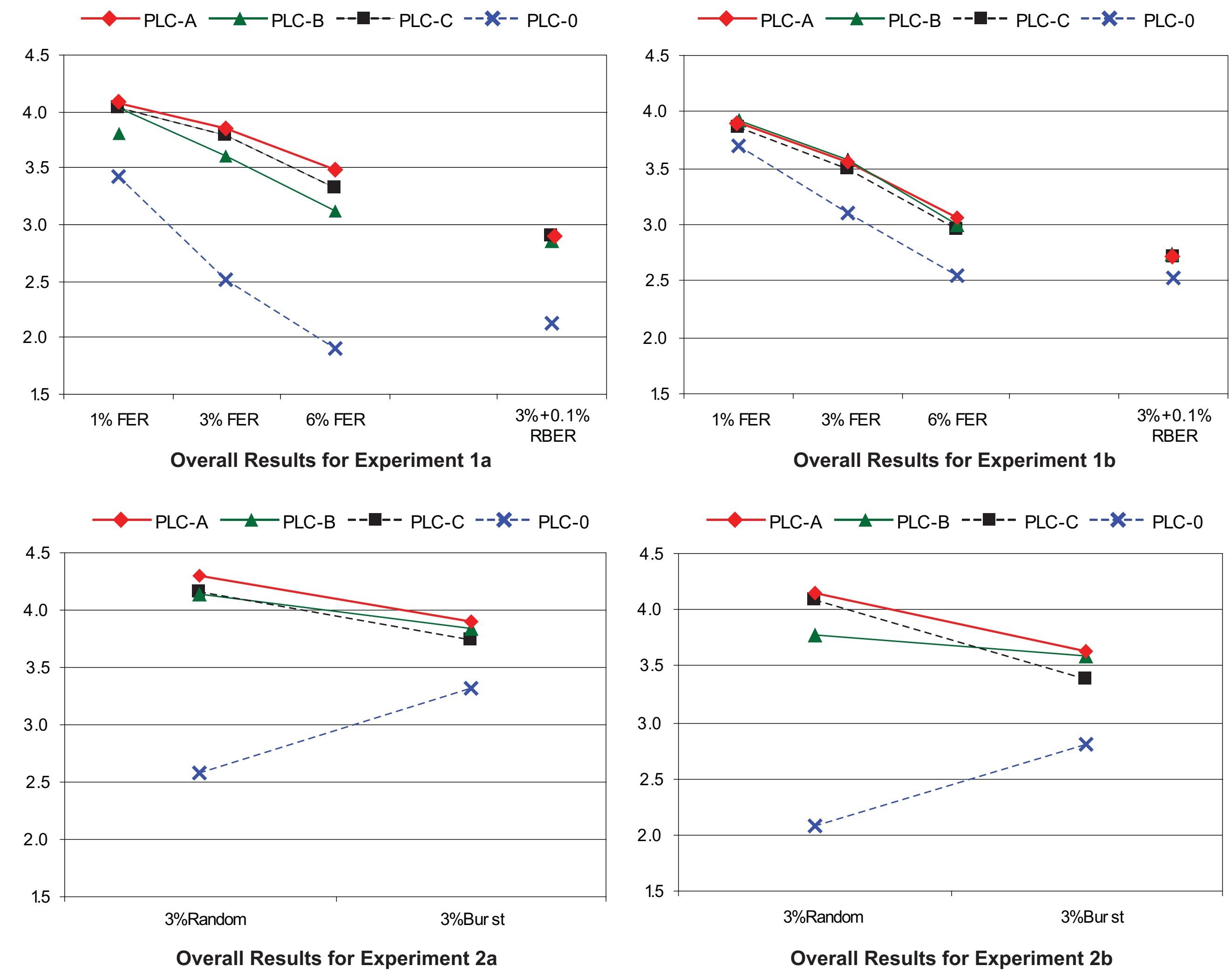


**Figure 3: ITU-T G722 III Tests**

# Reduced-Complexity BVC (RC-BVC)

Although the BVC algorithm discussed above already has a relatively low complexity, for some applications it is desirable to have even lower complexity. The Reduced-Complexity BVC (RC-BVC) was developed to address this need. It is derived from BVC with the goal of reducing the computational complexity and code size requirements of BVC as much as possible without degrading the output audio quality.

After many complexity-reduction techniques had been applied to BVC, the resulting RC-BVC algorithm achieved roughly 50% cycle count reduction compared with BVC, and the code size requirement was reduced by about 20%. The reduction of computational complexity and code size was achieved while maintaining essentially the same output audio quality as the original BVC.

## Low-Complexity PLC (LC-PLC) in SmartAudio

For some applications that require extremely low power consumption, such as Bluetooth headsets, even the lower complexity of RC-BVC may not be low enough. This is where Low-Complexity PLC (LC-PLC) comes in. Unlike many of the other PLC algorithms mentioned above that were derived from BVC or BV16/BV32, the LC-PLC algorithm was developed from scratch as an ultra-low-complexity solution using only extremely simple algorithm components. Some simple algorithm components from BVC/RC-BVC were re-used in LC-PLC, but the overall algorithm design is quite different and much simpler.

One particular emphasis of the LC-PLC design was to keep the average computational complexity (in addition to the worst-case computational complexity) as low as possible. Another emphasis was to minimize the code size as much as possible. The resulting LC-PLC achieves significantly lower average and worst-case computational complexity and code size than BVC or RC-BVC.

The LC-PLC algorithm can be applied to PCM directly or to Bluetooth standard codecs such as CVSD (Continuously Variable Slope Deltamodulation) and SBC (Sub-Band Coding) with minor modifications. Since most of the Bluetooth headsets for cell phone application use the CVSD codec, special techniques have been used to enable better integration of LC-PLC with CVSD. Despite its very low complexity, the LC-PLC manages to maintain essentially the same output audio quality as BVC at least for the main target application of the CVSD codec for Bluetooth headsets. Figure 4 shows the speech quality as measured by PESQ for a CVSD decoder with the ramp-down-to-zero approach, which is a conventional way for Bluetooth headsets to handle packet loss, and a CVSD decoder integrated with LC-PLC. For packet loss rates at or above 3%, the LC-PLC algorithm provides a PESQ improvement of 0.50 to 0.61. This is a huge audio quality improvement and it makes a difference that is audible to casual listeners. In fact, the third and last waveforms in Figure 1 are the output signals of the ramp-down-to-zero approach and LC-PLC, respectively, for the Bluetooth CVSD codec. The LC-PLC waveform shows a visually obvious improvement.

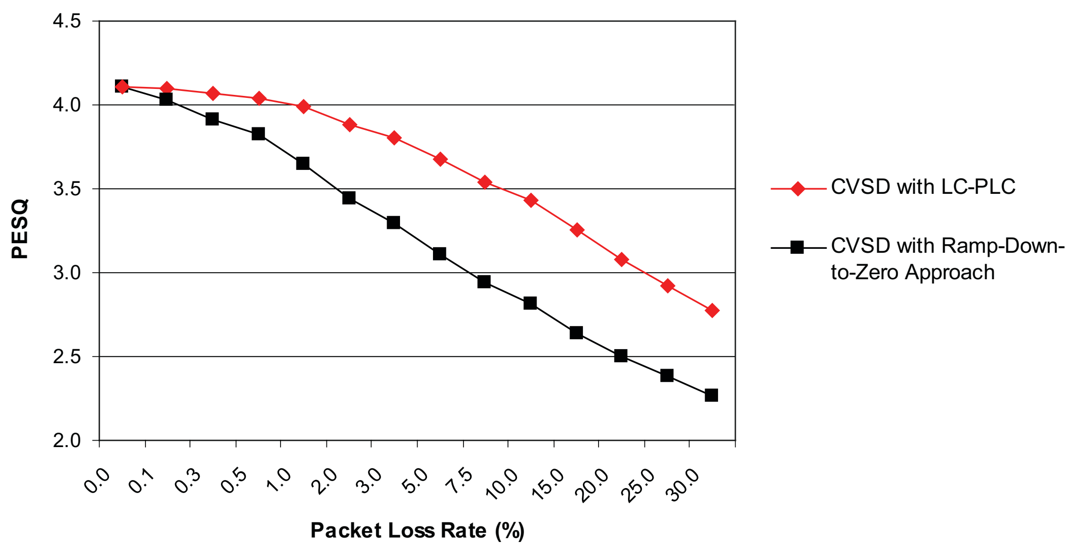


**Figure 4: PESQ Comparison of LC-PLC and Ramp-Down-to-Zero for Bluetooth CVSD Codec**

The LC-PLC algorithm has been ported to most of Broadcom's Bluetooth chips that handle telephone calls. Real-time hardware demonstrations of LC-PLC based on these Bluetooth chips have been shown to numerous potential customers, including cell phone manufacturers and Bluetooth headset manufacturers. Customers were universally astounded, and Broadcom was awarded many design wins partially due to the success of LC-PLC. The LC-PLC algorithm is part of the collection of Broadcom's SmartAudio algorithms specifically optimized for Bluetooth audio quality enhancement.

# Conclusion

As can be seen from the preceding discussions, Broadcom's PLC technologies have both the breadth and depth necessary to address a wide variety of application scenarios with very-low-complexity PLC solutions that offer superior performance and dramatic audio quality improvement. These PLC technologies lead the industry in performance, complexity-reduction, and features, and most are already ported to Broadcom chips and are readily available to help Broadcom's customers to reap the benefits of unrivalled audio quality improvement.

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**BROADCOM CORPORATION**
5300 California Avenue
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