iLBC White Paper -- October 15, 2004

iLBC – Designed For The Future

Introduction

Since the introduction of VoIP, there have been concerns regarding the current low bit-rate codec standards. The main issues impacting the manufactures of VoIP equipment and developers of applications involve the complex IPR management due to many patent holders, costly licensing models, and poor quality in real IP networks. In 2000, Global IP Sound decided to develop a codec that would meet the needs of the VoIP industry. The objective was to take advantage of the expertise within Global IP Sound to develop a codec that would be royalty-free, designed specifically for packet communication, offer high voice quality for both clean and packet loss conditions and bring it forward to different standardization bodies for interoperability compliance. The new codec is now available under the name iLBC.

History

Most contemporary speech codecs are based on the Code Excited Linear Prediction (CELP) coding paradigm. Examples of such codecs are ITU G.729 and G.723.1, GSM-EFR, and 3GPP-AMR. The CELP method has been shown to be very successful in achieving good quality at low bit-rates for circuit switched networks. The main reason for the effectiveness of this coding method is that it is based on exploiting the interdependencies present between consecutive speech segments. The performance of a CELP codec is hence heavily dependent on the history of previous encoding. The CELP coders are memory based and consequently error propagation results from lost or delayed packets. This obviously is a major drawback for packet-switched communication since the result is that a single lost packet affects the quality of many future packets.

iLBC, the codec

iLBC is a speech codec developed for robust voice communication over IP. It is designed for narrow band speech, with a sampling rate of 8 kHz. The iLBC codec supports two basic frame lengths, giving a bit-rate of 13.3 kbps with an encoding frame length of 30 ms and 15.2 kbps with an encoding frame length of 20 ms.

Deployment of the iLBC algorithm results in a speech coding system with a controlled response to packet losses. iLBC treats each packet independently from all other packets, making it ideal for packet communications. The codec displays a graceful speech quality degradation with increasing severity of IP packet loss and/or delay. This contrasts with the behavior of codecs based on the CELP paradigm, which were originally designed for circuit switched or wireless networks and designed to be resilient to bit errors instead of packet loss.

A relevant metric for a speech codec used in scenarios where packet losses are present is the number of frames/packets it takes to recover from a single packet loss. In the case of iLBC, this number is zero. The first packet following a lost packet is always decoded exactly as intended.

While iLBC is a narrowband speech codec, it utilizes the full 4 kHz frequency band available, whereas most standard low bit-rate codecs only use the frequency band from 300 Hz to 3400 Hz. This has a clear effect on speech quality. In addition, the spectral characteristic of speech coded by iLBC accurately mimics the characteristics of the original signal resulting in a much more natural and crisp sound than that from standard low bit-rate codecs.

To summarize, the iLBC algorithm enables state-of-the-art fixed bit-rate coding for packet networks with an excellent quality-versus-bit-rate tradeoff.

Standardization

In April 2004 iLBC was specified as a **mandatory** codec in the released CableLabs PacketCableTM 1.1 Audio/Video Codec Specification for multimedia terminal adapters (MTAs) and media gateways. “Since GIPS iLBC code is specifically designed for packet networks, we believe this level of specification provides cable operators the high performance and voice quality required for their VoIP solutions to gain an edge with their customer base,” said Steve Craddock, Senior Vice President of New Media Development at Comcast and Chairman of the CableLabs PacketCable business team.

The iLBC was accepted in March 2002 by the IETF as the first speech/audio codec to be standardized and today the iLBC codec is in the final stage of the standardization process in the Internet Engineering Task Force (IETF), as part of the IETF Audio Visual Transport (avt) Working Group.

Codec Performance

Several performance evaluations have been made by Global IP Sound and independent laboratories. In 2002 Dynastat Inc. performed a formal listening test of iLBC and in 2003 AT&T’s Voice Quality Assessment Lab also performed extensive testing of the iLBC codec.

The picture below depicts the results of the Dynastat evaluation, where the 30 ms mode of iLBC was benchmarked with the existing coding standards G.729A and G.723.1.The results clearly show iLBC's superiority when used in a real life environment, where its intrinsic packet network properties result in a high quality even under adverse network conditions.

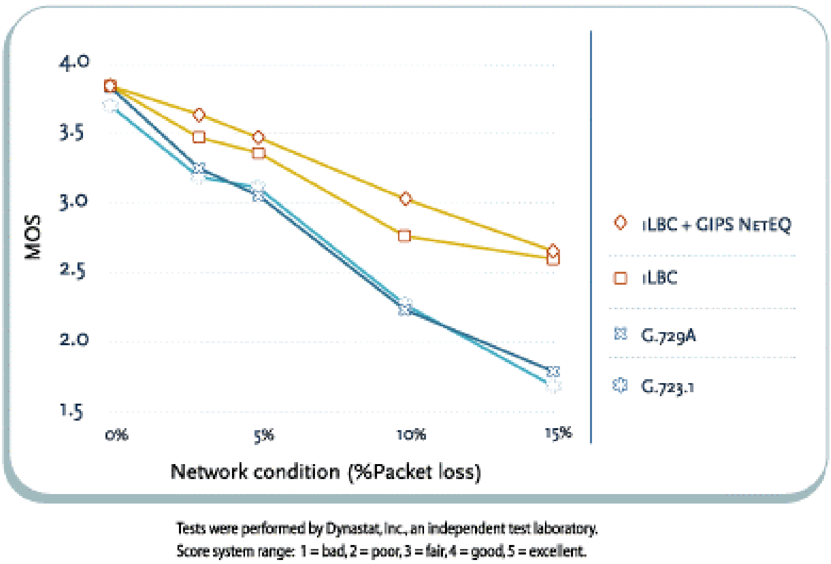

Figure 1. *Comparison of iLBC, G.729A and G.723.1 performed by Dynastat, Inc.*

The tests also showed that iLBC not only performs significantly better than current standard codecs (G.723.1, G.728, G.729, GSM etc) under packet loss conditions but also equal to or better than standard codecs in clean channel (no packet loss) conditions.

Condition: Clean channel				
ILBC is:	> G.723.1	>= G.729	>= G729.A	<= G.729.E

Condition: Packet loss				
ILBC is:	>> G.723.1	>> G.729	>> G729.A	> G.729.E

The AT&T tests also showed that there was no significant performance difference between the 20 ms and 30 ms modes in iLBC, except under packet loss conditions where the 20 ms mode showed even higher packet loss robustness. AT&T VQA Labs also stated that iLBC did well in the presence of background noises and performed comparable to G.729.E for a channel without packet loss.

Implementations

As of today, several manufactures of VoIP equipment and application have integrated iLBC in their products. In the list below we show a few examples of companies that are using iLBC in their commercial products:

- Applications/Soft phones: Skype, Nortel, Webex, Hotsip, Marratech, Gatelinx, K-Phone, XTen;
- IP Phones: WorldGate, Grandstream, Pingtel;
- Chip: Audiocodes, TI Telogy, LeadTek, Mindspeed.

Licensing iLBC

Device and application manufactures always search for cost-effective ways to meet new requirements and provide features to the market. Several aspects need to be considered when making the decision to either develop the iLBC code in-house or license the iLBC code from an external supplier.

Licensing iLBC code from an external supplier can provide substantial development cost savings, higher quality, time to market acceleration, lower risk and increased flexibility. The selection of the supplier should be made carefully to minimize any risk, or additional cost.

The selected supplier should:

- Posses the technical expertise necessary to perform the very sensitive implementation and testing issues related to porting iLBC to a fixed point DSP environment.
- Meet the technical requirements in terms of MIPS, quality, code size and memory for the selected platform(s).
    - Floating point (FLP) to fixed point (FIP) conversion is a trade-off between code efficiency, memory usage, and most importantly speech quality.
    - A poor implementation of iLBC FIP code will delay the porting to a DSP.
- Provide a track record in fixed point ANSI C conversion of speech codecs, DSP optimization and signal processing skills.
- Provide a track record with iLBC licensees for the selected platform(s).

Selecting a proven iLBC supplier is necessary to guarantee timely results and high quality performance.

Calculating in-house iLBC design cost

To calculate the design hours it takes for an experienced designer to convert the floating point code to fixed point ANSI C code or to a DSP platform, the following assumptions have been made:

- Work performed by a senior engineers with strong codec, FIP and DSP knowledge;
- "Standard" optimization (most of the code in C, critical parts in assembler language);
- DSP conversion is based on high quality iLBC FIP code.

Design hours for converting the iLBC code in-house	Effort
Floating to fixed point conversion estimate	4-6 man-months
Fixed point to DSP conversion estimate	3-4 man-months

The table above only considers the design effort. To get a full picture of overall cost, the following aspects must also be considered:

- Internal training;
- Lead-time to familiarize with the iLBC codec;
- Project risk – complexity, performance, and quality can be not be verified beforehand, as when iLBC is licensed from external supplier;
- General technical and business risks;
- Documentation;
- Testing;
- Support and life-cycle costs;
- Overhead – the fixed costs associated with staff, tools etc.;
- Unavailability of engineering resources for the development of other products and / or features.

In addition, the value of a short time-to-market should not be underestimated. By licensing the optimized iLBC code, the final product can reach the market many months earlier than if the code was to be developed in-house.

iLBC code availability

Floating point ANSI C (FLP):

The specifications of the floating-point iLBC and ANSI C code can be downloaded via www.iLBCfreeware.org or www.ietf.org sites.

Fixed point ANSI C (FIP):

The Global IP Sound iLBC FIP code has been delivered to several customers and the quality has been verified to be indistinguishable from the floating point code.

DSP/MIPS:

At the time of this writing, Global IP Sound has already provided our customers well-optimized iLBC code for the most common platforms on the market. The usage of GIPS implementations avoids surprises when using the iLBC-optimized code.

About Global IP Sound

Global IP Sound develops embedded voice processing technologies for real-time communications over packet networks. GIPS SoundWare™ provides better than PSTN voice quality and fidelity in end-to-end IP communications with robustness against packet loss. Global IP Sound's world-renowned speech processing and IP telephony experts deliver these solutions to applications developers, gateway and chip manufacturers. Companies using GIPS SoundWare products include Nortel Networks, Skype, WebEx, Marratech, and other key players in the VoIP market. Global IP Sound is a member of the Intel® PCA Developer Network, the Motorola Design Alliance, and Texas Instruments' third party developer network. Global IP Sound has headquarters in Stockholm and San Francisco. More information is available at www.globalipsound.com.

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