

Engineering Note

EN0805-009

Coral Sea Softswitch & Coral VoIP Gateways Quality of Service

Overview

Coral Sea is a powerful softswitch that provides unified communication in conjunction with multimedia over IP.

The Sea softswitch is the cornerstone for NGN networks and is used as an applications platform to comply with the needs of the organization.

The concept behind the Sea softswitch is twofold – on the one hand, it is designed to preserve a company's investment by retaining current telephony equipment; on the other hand, it allows businesses to migrate their telephony network in order to provide multiple communications services over heterogenic networks.

With the Sea softswitch, employees, customers and business partners can communicate easily using advanced unified applications, including:

- Unified messaging
- Multimedia conferencing (Voice and Video)
- Multi-Layered Mobility (MLM) utilizing softphones, dual-mode phones and IP phones
- Presence

Top Voice Quality

Tadiran presents an innovative VoIP solution, which complies with demanding requirements for best Voice quality, QoS (Quality of Service) and GoS (Grade of Service).

Utilizing sophisticated algorithms and powerful hardware-based DSP (Digital Signaling Processor), Tadiran meets these challenges while effectively handling network service impairments.

Bandwidth-saving methodologies

Tadiran's VoIP solution utilizes:

- Media compression
- RTP header compression (Router level through the WAN)
- Multiple frames per RTP packet
- Silence suppression mechanisms using Voice Activity Detection (VAD)

Media Compression Methods

Different compression methods can be configured according the network condition and bandwidth availability.

Available compression methods:
G.711A, G.711u, G.723.1_5.3, G.723.1_6.4, G.729A CS-ACELP

The Tadiran solution negotiates the best match of its available high quality compression methods to current network and traffic conditions. Tadiran's dynamic bandwidth utilization assures maintenance of top voice quality in spite of varying network conditions (which may include high packet loss, jitter, etc).

Multi-Frames per RTP Packet

Minimization of RTP overhead is essential to making the widest possible Bandwidth available to customers. Coral Sea Port can be configured to define the number of multi-frames per Media packet in order to keep RTP overhead as low as possible. When the number of multi-frames per Media packet is large, the probability of packet loss is reduced at the cost of greater network latency. High latency may influence the overlap phenomenon and may cause problematic echoes over the network. In order to meet the challenges of high voice quality and carrier class QoS, Tadiran uses a sophisticated Echo cancellation mechanism that complies with G.168 (Extended Echo Cancellation), and an efficient Echo tail mechanism (up to 128 ms). The Tadiran solution requires less bandwidth and is less vulnerable to bandwidth congestion. These advantages are the results of using longer packets (including large payloads) and the consequent reduction in the RTP packets rate.

Silence Suppression Mechanisms

Tadiran's solution enables the activation of silence suppression mechanisms based on Voice Activation Detection (VAD) mechanisms. Silence suppression results in more efficiency and an overall reduction in bandwidth use. Using Tadiran's VAD mechanism, voice quality is not affected by possible high frequency background noise, front-end clipping or back-end clipping. During noise suppression, the comfort noise generator is activated to ensure that voice naturalness is un-affected by the noise suppression.

Delay

Delay is a common IP environment impairment. Talker overlap becomes significant when latency increases to more than 250ms (one-way delay), and high latency influences perceptual quality, with phenomena such as echo becoming a major problem. In order to limit latency and assure superior performance, Tadiran uses non-blocked powerful resources such as hardware-based DSP for efficient compression, VAD, DTMF detection, and an RTP Packet-structure controller.

Echo

Echo phenomenon is measured in several frequency ranges:

- SRL-Low: 300-500 Hz
- SRL-High: 2200-3400 Hz
- ERL: 560-1965 Hz

Tadiran's built in echo cancellation mechanisms comply with the G.168 standard and successfully handle two important aspects of the echo phenomenon: echo tail and return loss.

Echo tail is the amount of time that passes from the sending of media until correlated media is received at the destination. Tadiran's cancellation mechanism treats correlated media in a predefined interval that is shorter than the echo tail (in ms).

Return loss is the echo attenuation, i.e. the ratio between the transmitted signal and the received echo (in dB).

JITTER

Since packet switches are connectionless, the one-way delays of packets vary.

Variations in inter-packet delay result in the jitter phenomenon. In real-time applications such as voice, the jitter phenomenon requires treatment. According to RFC 1889, the 'relative transit time' of two sequential packets i and $i-1$ (expressed as D_i) is defined as:

$$D_i = (Tr_i - Tr_{i-1}) - (Ts_i - Ts_{i-1}) = \underbrace{(Tr_i - Tr_{i-1})}_{OWD_i} - \underbrace{(Ts_i - Ts_{i-1})}_{OWD_{i-1}}$$

$$Jitter(n) = \frac{\sum_{i=1}^n |D_i|}{n} ; n = 16$$

OWD- One Way Delay

Where Tr is the absolute timestamp of the receiver clock and Ts is the RTP packet timestamp of the sender.

In order to overcome jitter, the system should include a jitter buffer. The length of the jitter buffer may prevent packet loss and increase the ability to control a constant media rate. However, there may also be an undesirable effect on latency. The longer the jitter buffer is in use, the higher the latency that will be imposed into the system.

To optimize jitter-buffering, Tadiran's system uses an adaptive dynamic jitter buffer with the following features and characteristics:

- Individual engagement of multiple channels
- User-tuned dynamics and tradeoffs
- Suitable for both delay sensitive and error sensitive applications

The application controls the operational mode of the jitter buffer and the way it responds to network behavior by its selection of the available optimization factors. These are determined in light of application objectives and the network mode.

If the optimization factor is set to the maximum value, the jitter buffer delay tracks the network latencies to their maximum and stays there, thus minimizing the packet error ratio but causing high voice path delay. If the lowest value is used, the jitter buffer increases the delay only to compensate for clock drifts, reverting each time to the minimal setting. This minimizes the delay at the cost of higher packet loss rates.

Congestion

In order to minimize Congestion phenomenon, which may lead to packet loss, Tadiran's system works in conjunction with existing routers (on the VoIP network) that include built-in QoS mechanisms such as MPLS, DiffServe, RSVP, etc.

By tagging the RTP packets using TOS (Type Of Service) bits, the system can define packet priority and the proper treatment by the QoS mechanism.

Then, by matching the length of the packets to the network conditions, using some of the methods already mentioned, e.g. multi-frames per packet, Voice Compression, etc., Tadiran's solution minimizes the congestion phenomenon.

QoS and Voice Quality factors

- Front-end or back-end clipping: efficient VAD
- Minimization of packet loss
- Dynamic jitter buffer
- Priority or congestion problems over network (in conjunction with the routers)
- Echo cancellation compliance with G.168
- Effective echo tail – up to 128ms
- Reduction of background noise aimed at maintenance of top voice quality using effective VAD
- Treatment of cross-talk
- Reduction of latency: packets structure controller, effective dynamic jitter buffer
- High performance hardware DSPs for voice compression
- Effective embedded handling of harmonic distortion (for 2nd & 3rd harmonic of 1004 Hz at 0 dBm)
- Effective embedded mechanism to handle any Quantization distortion