PACE

RTP Performance Enhancing Proxy

V2

Revision history table if required.

Document Issue	Date	Inmarsat Owner	Notes
V1	21/01/2013	Lee Marston	
V2	12/02/2013		

Contents

1. BASICS IP Router- What's it about.

BASICS is the essential toolbox, offering a simplified approach to the deployment of networking, voice and IP solutions.

Each BASICS device provides a number of ports for connection – either to an IP WAN such as a satellite modem or over a local fibre optic connection for local reach back. Designed for applications where efficiency is critical – power, space or bandwidth – BASICS provides specific solutions for every day challenges in communications deployment.

Each BASICS unit offers a single, simple primary mode of operation, such as IP Routing, Optimisation, Voice (including STE / STU support), Four-Wire or Radio Relay, and all can be software enabled with network router functionality too.

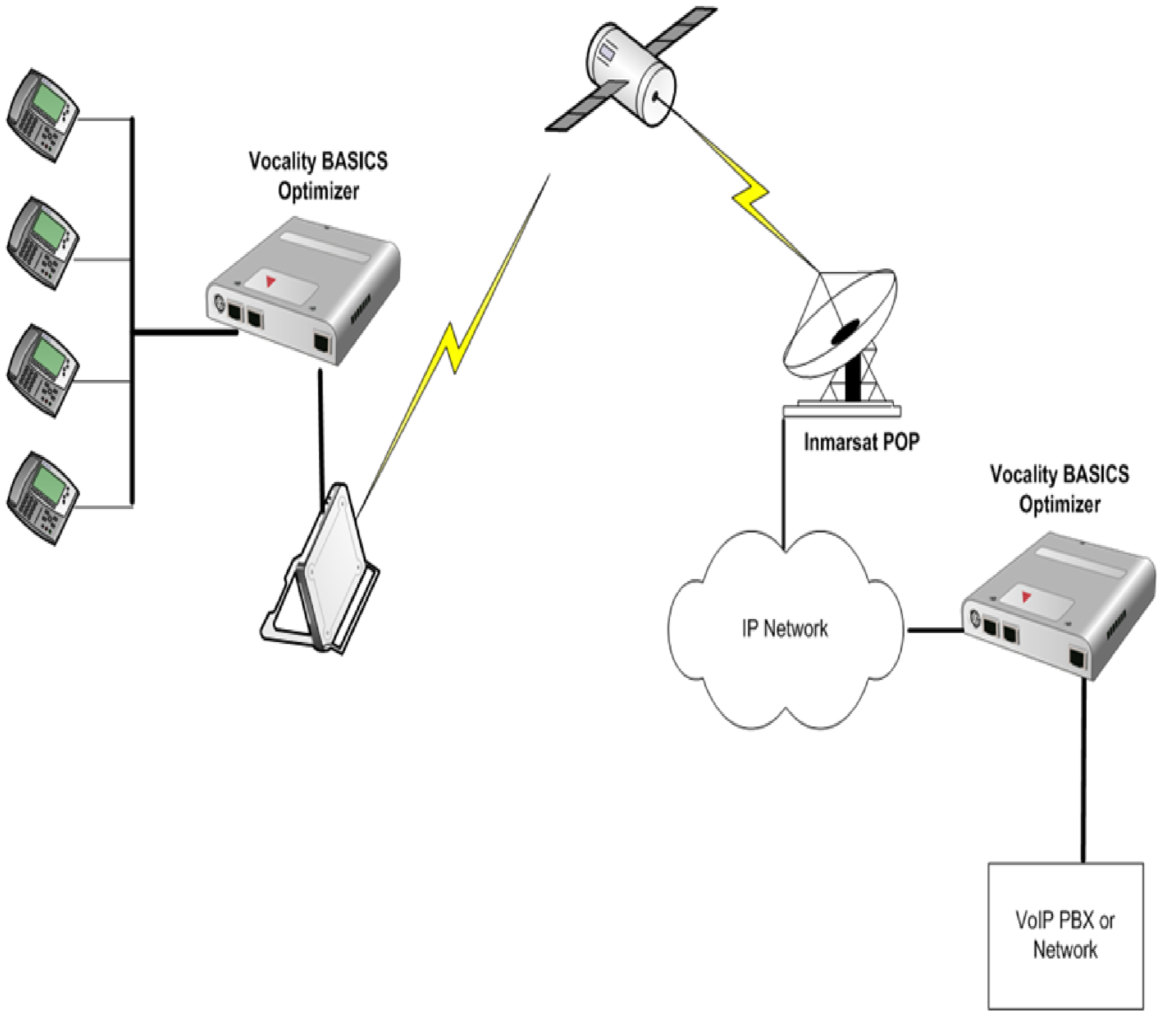
All the BASICS platforms can be delivered in one of the following formats, ensuring both standard product network operators and tactical systems integrators have a form factor that meets their requirements.

2. Real time Protocol (RTP) issues, VOIP problems.

VoIP, RTP and other real time IP applications can suffer when used over an IP based satellite system due to propagation latency and jitter, and on terrestrial internet style connections where no guarantee of quality of service is delivered.

3. The Vocality Solution

Improves quality of voice over IP conversations across contended lossy networks with large jitter, reducing the bandwidth required to tunnel these conversations across the IP network.



4. Feature Summary

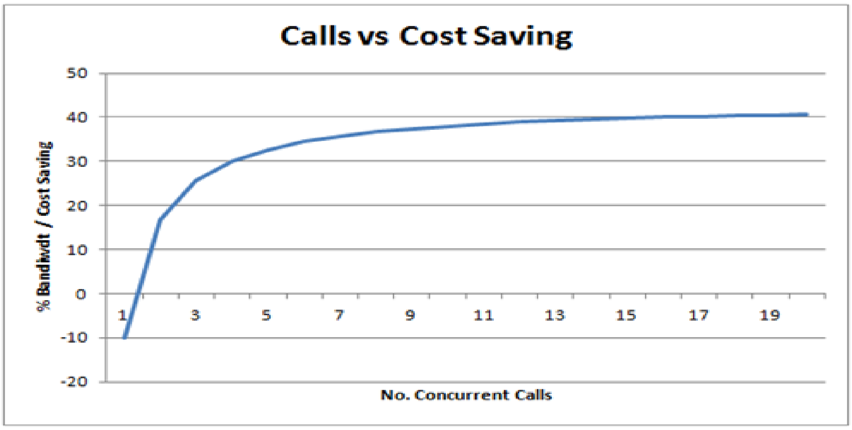
The Vocality performance enhancing proxy for Voice over IP traffic involves five different features designed to enhance the performance of VoIP and RTP over IP networks:

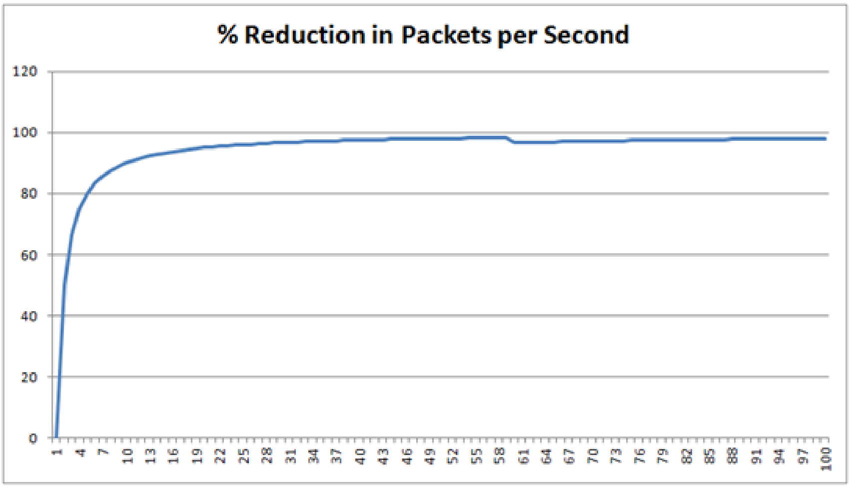
- Header compression for RTP traffic
- SIP SDP 64k codec filtering
- Jitter buffering for RTP traffic
- G729B small sample filtering
- Packet aggregation

5. PACE Technical Detail

5.1 Packet Aggregation

Vocality uses packet aggregation, putting many smaller packets into one packet, across its WAN aggregate link. This minimises the IP overhead and headers.





5.2 Header Compression for RTP Traffic

Portions of the IP, UDP & RTP headers are constant throughout an RTP session (conversations). The Vocality unit can find these conversations, inform the IP peer of the contents of the headers, and then send only a conversation identifier. When the compressed packet arrives at the target IP peer, the headers are reconstituted and sent to the RTP target. RTP sequence and timestamp information is forwarded intact. So each RTP packet appears on the aggregate with 32 fewer bytes.

Each packet to be sent to the peer is examined to identify valid packets (IP & UDP checksums) and known conversations. Any UDP traffic with an even port number (except the SIP port number 5060) is considered RTP traffic. Service management filters ensure that only RTP port numbers used in the target network are forwarded down the RTP compression enabled channel.

5.3 SIP SDP 64k codec filtering

The SIP SDP 64k codec filtering logic looks for session description protocol messages within SIP signalling messages. It strips out any PCMU and PCMA (G711 64k) codec negotiation to prevent SIP devices selecting 64k codecs to use over the network.

5.4 Jitter Buffering for RTP Traffic

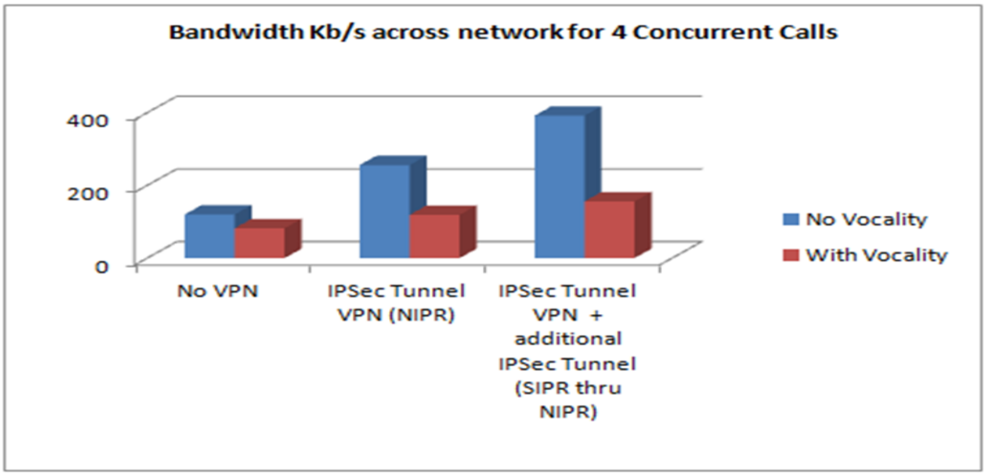
Some SIP and RTP devices cannot tolerate the large jitter seen in some IP satellite networks. The Vocality unit seeks to remove this jitter from an RTP stream that is forwarded through the embedded IP router and across the satellite network.

RTP packets that are forwarded from the IP router to an IP tributary are pre-pended with a timestamp. This timestamp is sent across the satellite network with each RTP packet, and the timestamp is used at the peer unit to forward packets onward at the same frequency with which packets arrived.

Note: This relies on timestamps from the CPU clocks on peer units across the Vocality network and using these to control packet synchronization

5.5 G729B Small Sample Filtering

When a G729B codec sends data over an RTP stream it typically generates 20ms samples of 20 bytes (plus RTP/UDP/IP overhead). During "normal" conversations some G729B devices generate smaller samples (of 10ms) as voice/silence transitions occur. Several of these may occur per second during "normal" speech. If these voice samples are not forwarded, the perceived voice quality is not greatly affected but the bandwidth required to forward them is saved



6. Benefits to BGAN users

- Vastly improves VoIP call quality when using BGAN Background/Standard IP.
- Reduces cost by removing un-necessary packet overheads.
- Simple connection to BGAN.
- Fail to wire.

7. Further Details and Support

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