NETRIX

VoIPAK & VoIPZIP

VoIP Network Optimization

VoIPAK and ***VoIPZIP*** from NSGDatacom Inc. are powerful packet optimization and bandwidth compression products for VoIP SIP and MGCP trunking applications.

VoIPAK is a packet optimizer that reduces the packet overhead associated with multiple VoIP calls traversing common network connections.

VoIPAK combines the packet streams from multiple VoIP calls into a single packet stream, typically reducing the number of network packets per second (pps) by a factor of 50:1 or more, and reducing the bandwidth required for trunking calls over common network connections by up to 3:1.

For Example ***VoIPAK*** can reduce the typical network load of 50 simultaneous VoIP calls from ***5000 pps to only 100 pps***. Since only one voice sample from each call is placed in each packet no noticeable delay is added to any of the calls being transported.

By this means ***VoIPAK*** also eliminates the significant IP overhead normally associated with VoIP calls. In the above example the total IP bandwidth required to support 50 G.729 VoIP calls is reduced from approximately 1.64 Mbps in each direction to less than 450Kbps in each direction, ***a resulting bandwidth reduction of over 70%***.

Note that VoIPAK does not negatively impact the audio quality of the VoIP calls because the audio payload is transported in its entirety across the network. In fact the audio quality is normally improved substantially due to a significant reduction in network related packet loss when using ***VoIPAK.*** ***VoIPAK*** units works in both point to point and fully meshed modes and operate transparently to users at all times.

VoIPZIP Is the ***VoIPAK*** packet optimizer with the addition of integrated voice compression.

VoIPZIP provides all the functionality of ***VoIPAK*** and also compresses the voice payload of G.711 VoIP calls using one of several optional compression formats. ***VoIPZIP*** eliminates the considerable throughput bottlenecks often associated with trunking uncompressed (G.711) VoIP calls over conventional wireline, satellite or wireless network connections.

For example, most service providers find that a single 1.544Mbps (T1) data connection typically supports no more than 15 standard G.711 based VoIP calls before call quality is compromised. 15 standard uncompressed VoIP calls use 1.2Mbps of bandwidth in each direction and generate 1500pps. Using the packet optimizing techniques of ***VoIPAK,*** the packet rate generated by 15 VoIP G.711 voice calls is reduced from 1500pps to 100pps. However, due to the uncompressed voice content, the G.711 based audio still uses approximately 970Kbps of bandwidth in both directions.

With the additional voice compression capabilities of ***VoIPZIP*** the audio content of G.711 VoIP packets is compressed using one of our standards-based compression engines. The resulting bandwidth required to support 15 toll quality voice calls (typical MOS of 3.9) is only 130Kbps in each direction. The optional use of silence suppression reduces the required bandwidth by a further 40%-50%, ***to 80Kbps or less,*** resulting in a total ***bandwidth saving in excess of 90%.*** With our optional low bit rate compression (typical MOS 3.7/3.8) ***a bandwidth reduction of 16:1 can be achieved.***

VoIPAK and ***VoIPZIP*** are highly flexible, configurable networking platforms with many additional benefits and optimization features not covered by this application note. Optional integrated WAN ports provide additional performance benefits over Ethernet connections. Graphical performance examples shown overleaf are typical for IP and can be exceeded in some applications.

VoIPAK and ***VoIPZIP*** are designed for use in Carrier grade networks and are fully supported by the ***NetrixView*** Network Management System. The NMS interface provides comprehensive GUI support for remote configuration, diagnosis, statistical call analysis and other management functions. A range of ***VoIPAK*** and ***VoIPZIP*** platforms are available for CPE and Central Office applications which are fully interoperable with other products in the ***Netrix Network Exchange*** product line.

Netrix brand products are Installed in many mission critical networks worldwide, and continue to provide dependable voice and data transmission in carrier networks, call centers, military, transaction processing, financial, airport, service provider, and other enterprise applications.

Communication solutions from

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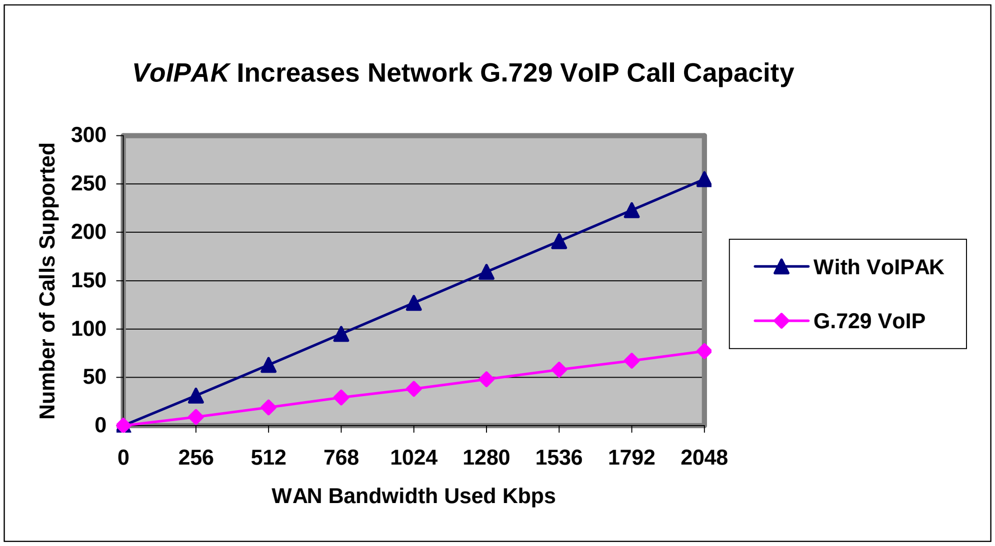
Performance Data

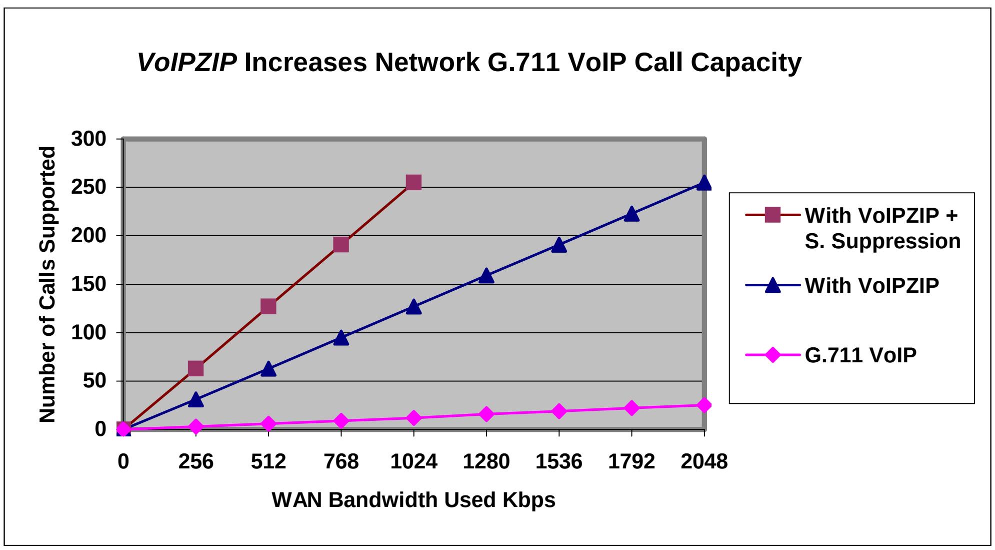


VoIPAK increases the number of G.729 (and other low rate codec) based VoIP calls supported on a network connection by reducing IP packet overhead.

Example: Using ***VoIPAK*** bandwidth utilization is improved by a factor of 3.3:1.

Note: G.729 curve does not show any VoIP saturation due to high packet throughput which could cause VoIP calls to be limited, further enhancing the value of ***VoIPAK*** In this example.

Assumptions:
G.729 Sample size 20 Bytes (default)
IP (UDP/RTP) Headers 40 Bytes
MLPPP or Frame Relay Header 6 Bytes

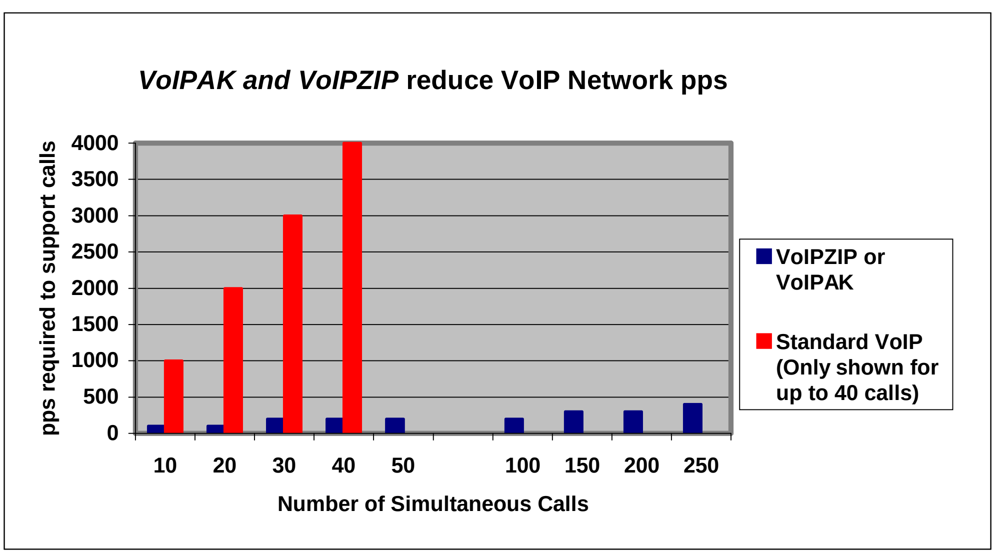


VoIPZIP increases the number of G.711 based VoIP calls supported on a network link by compressing voice content and reducing IP packet overhead.

Example: Using ***VoIPZIP,*** bandwidth utilization is improved by a factor of 10:1 without silence suppression, and 20:1 with silence suppression.

Assumptions:
G.711 Compressed to 8Kbps voice.
Silence suppression assumes 50% silence in each direction.

In the above Graphs WAN Bandwidth Used is full duplex, actual transmitted data in both directions.



VoIPAK and ***VoIPZIP*** can typically reduce the rate of VoIP network packets by a factor between 30:1 and 70:1 depending on the platform and its configuration.

Assumptions:
Sample rate is 20ms, (50pps each way).
pps shown is total sum of inbound and outbound packets.

Multiple input packets from a single VoIP call can be in the same output packet for greater packet efficiency (not illustrated).

VoIPZIP output sample rate can be configured independently of the input packet rate when compressing G.711 VoIP.

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