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# VoIP Bandwidth Utilization and Packet Handling

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TECHNICAL WHITE PAPER

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## Introduction

Despite long-standing forecasts that bandwidth will become essentially free and unlimited, the practical reality is that bandwidth utilization remains important for most voice over IP (VoIP) providers. Getting trunk bandwidth to points of presence (POPs) for thousands of calls can quickly become expensive.

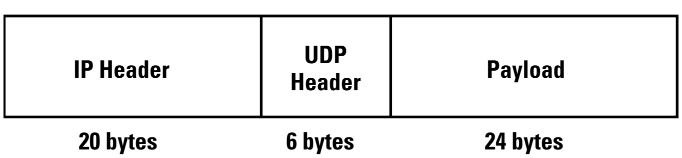
In packetized voice communication systems — VoIP, frame relay, or ATM — voice data is digitalized and lossy compressed into frames. Each frame represents the voice data for a small unit of time, typically 30 milliseconds. Frames are then transported over the network from a source to a destination. The frames are then decompressed.

In a VoIP network, each voice data frame is typically encapsulated in one datagram. The Internet Protocol (IP) imposes a minimum of 20 bytes of header, containing such information as the destination IP address. The User Datagram Protocol (UDP), typically used for voice transport applications, adds another six bytes of header information. For example, a voice frame encoded with the G.723.1 voice algorithm at 6.4 kbps is 24 bytes long.

This results in a total packet length of 50 bytes, of which 26 bytes are overhead. Since 52 percent of the bandwidth is effectively wasted by the large amount of overhead the Internet Protocols impose on the packets, this greatly reduces the number of calls that can be supported at one time. For example, a call compressed using the G.723.1 at 6.4 kbps codec would require 13.3 kbps of IP bandwidth to transport the 6.4 kbps TDM call.

**Figure 1: VoIP Packet**

In a typical VoIP packet, the IP and UDP header can account for the majority of the data in the packet.



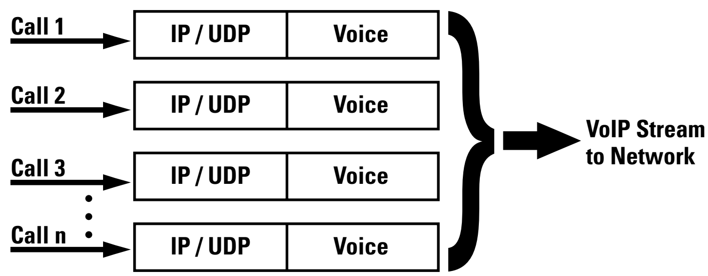

## VoIP Packet Processing

the router capabilities connected to the VoIP node. To have low latency, voice packets need to be very small. Many voice circuits mean many small packet streams. The result is that the router connected to the VoIP gateway can be overrun by too large a volume of small IP packets. A router has bandwidth constraints, but it is sometimes overlooked that it also has processing power limits. Handling a large volume of small packets can require too much processing power from the router.

In the example of a voice call compressed with G.723.1 at 6.4 kbps, about 33 packets per second are generated. For 160 calls, more than 5,000 packets per second are generated! At that rate, a Cisco 2500 series router would be overwhelmed.

**Figure 2: Illustration of High Volume of Small VoIP Packets**

A traditional VoIP gateway sends many streams of very small packets to the network.




## Packet Grouping Multiple Frames from Same Call

One solution attempted by some VoIP equipment manufacturers is to group multiple frames from a call together in a single packet. The objective is to increase the amount of payload for a given header length. This approach has a significant disadvantage. If the equipment is configured to group five frames together before transmitting one packet, the latency end-to-end is increased fivefold. In the case of 30 milliseconds (ms) voice frames, since every packet contains five 30 ms frames, a packet can only be transmitted every 150 ms. Grouping smaller numbers of frames together reduces latency, but again increases the overhead.

**Figure 3: Packet Grouping Multiple Frames from Same Call**

Packeting multiple voice frames from a call into the same packet cuts down the IP overhead, but increases latency on the call.

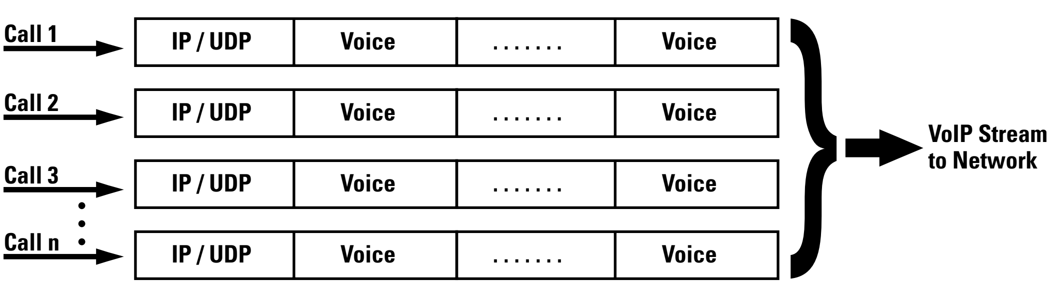
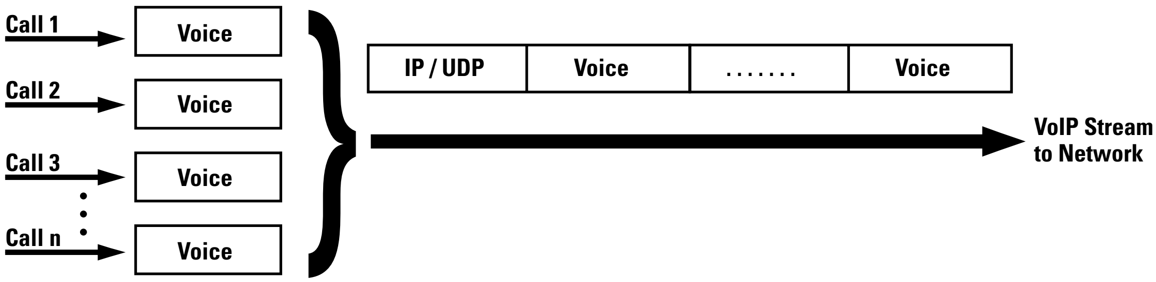


**Figure 4: Packing Multiple Calls into the Same Packet**

The system of frame packeting does not increase call latency, but ensures very low IP overhead as well as dramatically reducing the number of packets sent across the network.




A further disadvantage of this method is that if one packet is lost due to congestion or other network problems, a noticeable click, pop, or dropout will occur on the line. The average VoIP packet size from the above packing scheme is also not significant enough to greatly impact the performance requirements of the attached router.

## New System of Frame Packing

A new system of frame packing has been introduced that greatly increases bandwidth efficiency without impacting system latency. In addition, this new system addresses the issue of too many small packets overloading the router.

The new technique combines call time slices from the same moment of time into a different packet, improving efficiency without increasing latency. In net.com's SHOUTIP™ open telephony platform, as many as 60 calls can be combined into one packet. The IP header overhead is spread out over many calls and appears as a near negligible amount per call. No latency is added to each call as there is no delay in the transmission of each voice stream packet. Only 33 packets per second result from the 60 calls — two magnitudes less than the 2,000 packets per second required without frame packing.

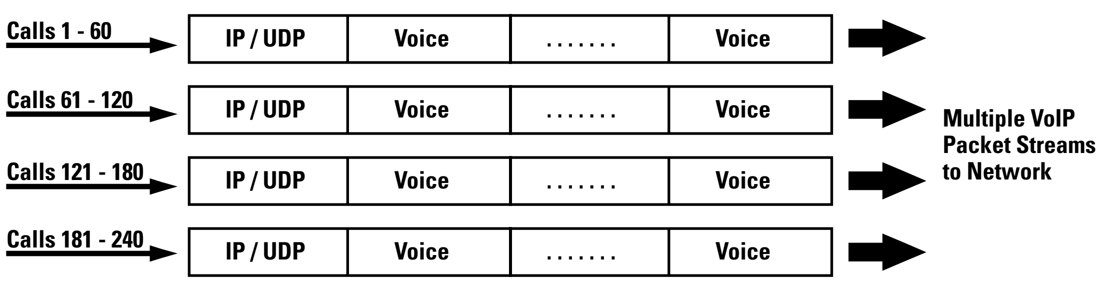
## System Architecture

Frame packing requires a system architecture that allows all calls within a node going to the same destination to be grouped together. Some VoIP architectures lack a central intelligence to do this, and are unable to coordinate calls across different voice boards. Other architectures even require an external system to implement frame packing. Not a very good solution for customers looking for fewer systems and a less clumsy network solution.

SHOUTIP architecture provides the best VoIP solution possible. All calls from the node are processed together such that all calls going to the same destination can be packed together. No external equipment is required to provide the voice packing algorithm. It is all contained in the node.

**Figure 5: Architecture of Inefficient VoIP Platforms**

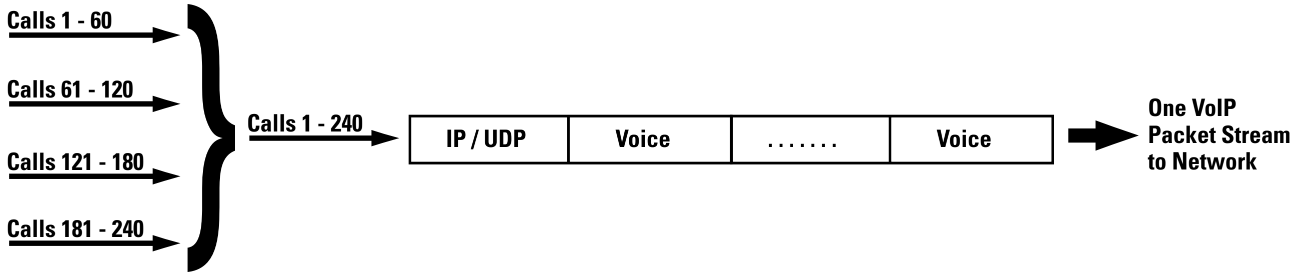
Some VoIP architectures limit the multiple stream frame packeting to one card only. Only calls originating from the same card and going to the same destination can be packed together.




(continued on reverse)

**Figure 6: SHOUTIP Architecture**

SHOUTIP considers all calls within a node for frame packing, no matter which care the call comes from. Hence, increased efficiencies are achieved by ensuring calls with the same destination are packed together, regardless of which interface card the call is on.




## Summary

VoIP offers tremendous opportunities for telecommunications services, but two key performance factors need to be considered for successful implementation. They are bandwidth utilization and packet handling. While in some parts of the network there is sufficient bandwidth, it can be an expensive commodity. Further, poorly designed VoIP equipment can congest a data network with a flood of minipackets that overwhelm access routers and generally slow down network performance.

The SHOUTIP platform offers a powerful VoIP solution for service providers. Available in all SHOUTIP's platforms is net.com patent-pending BESTflow™ frame packing technique. As suggested by the acronym BEST – Bandwidth Efficient Streaming Technology – BESTflow is a more efficient technique for "frame packing," or distributing and transporting voice over IP (VoIP) calls as data packets. With BESTflow, the SHOUTIP platform delivers an exceptionally low VoIP bandwidth overhead on top of the actual voice data. In addition, route requirements for packet handling are reduced by a hundredfold — adding to the savings when setting up a network.

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