VoIP Primer

4/25/2014

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NSG VoIP Primer

1. Overview

Voice over IP (VoIP) has been around for many years now. Telephone and cable companies offer VoIP services that are mostly indistinguishable in quality from the Plain Old Telephone System (POTS). Companies such as Vonage and Skype offer alternative high quality VoIP that is very cost competitive, so how can NSG sell it's specialized VoIP products into such a competitive market environment and compete with the major service providers?

The answer is threefold:

- **Firstly, we provide enhanced performance when IP throughput is constricted.**
- **Secondly, we provide enhanced performance when bandwidth is expensive.**
- **Thirdly we provide enhanced capabilities over conventional VoIP services.**

With regard to IP throughput, when the bandwidth available for placing calls is high, and there are no major delays or packet throughput limitations then standard VoIP systems work perfectly. For example, high bandwidth services designed to support video streaming can support VoIP without much of a problem. When abundant resource is available there is no discernable difference between VoIP calls and high quality telephone calls over POTS lines.

However, there are many circumstances under which IP bandwidth may be limited or, if not limited subject to random breaks in service, or if neither of these, very expensive. An obvious example is any transmission over a Satellite network. Another is transmission over Cellular 3G/4G data networks. Furthermore, all Internet based services drop IP packets more often than they might care to admit, so any solution that improves quality of service or provides enhanced capabilities by minimizing the impact of lost packets can offer an advantage.

The performance advantages of NSG (***Netrix*** brand) VoIP products are described below with direct reference to the IP network characteristics that affect the quality of VoIP telephone calls. Methods by which NSG products can enhance IP based services are described and the expanded capabilities of NSG VoIP products highlighted. **For example, NSG VoIP products are able to make reliable real-time fax and dial modem calls over IP networks where standards based VoIP equipment repeatedly fails.**

Also described are the enhanced functions and unique capabilities of NSG VoIP products that extend their application beyond the normal confines of VoIP systems. In particular, the capability of NSG products to optimize capacity and improve quality of service on standards based VoIP trunks, whether carrying few or many calls, and to combine these with other IP related services over a bandwidth-restricted channel.

2. VoIP Network Requirements

There are at least four network performance requirements that need to be met for VoIP telephone calls to be successful.

The first network requirement is the availability of adequate constant, uninterrupted bandwidth. Voice calls are real time and any network that 'drops' packets, even if the bandwidth is high, may lead to unacceptable VoIP quality. Even a short break in network availability can render a service unusable for telephone calls even though the IP service might be considered good enough for streaming video, since video can normally be buffered well ahead of when it is needed.

The second network requirement is for minimal network delay (latency). Telephone calls placed over IP networks depend on a constant steam of IP packets containing the audio. Although any noticeable latency across the network can be annoying, most people understand and accept the necessity of the delay when talking over a satellite link or even between cell-phones. Users may be willing to accept this, but any significant latency will prevent appliances such as fax machines and dial data modems from operating over a VoIP network.

The third network requirement is for minimal ***changes*** in the latency across the network – known as jitter. The receivers use buffering at each end of the link to absorb any jitter. In order to accommodate network jitter the buffer depth at the ends of the link is normally set to be slightly greater than the longest delay expected across the network. To set the jitter buffer any shorter would be to accept that some IP packets containing audio data will arrive late, causing an inevitable, (and possibly frequent) break in the audio output. Short breaks in the audio may not be noticeable to the human ear but will terminally corrupt data from fax transmissions or dial modems. Long or repeated breaks in audio will also render the VoIP quality unacceptable to users as well.

The fourth network requirement is for network nodes to meet a minimum IP packet throughput. Some older devices have limitations on packet throughput that can affect VoIP systems, especially if a large number of simultaneous VoIP calls are in progress. A typical VoIP access device divides each audio stream into 10ms sample periods, which generates 100 IP packets per second (pps) in each direction (200pps in total per voice call). A VoIP switch supporting just 10 simultaneous voice calls is therefore processing upwards of 2,000pps which can easily exceed the capacity of some network devices, particularly those used in some established satellite systems.

All the above requirements must be met for a network to support VoIP to a level comparable with a high quality POTS call. Clearly there are tradeoffs between these parameters to be taken into account when designing a VoIP system and a number of different techniques are employed to optimize these.

3. Standard VoIP Calls

Conventional telephone systems worldwide use a digital trunking system that converts POTS voice calls into a digital data stream at a bit rate of 64Kbps according to the G.711 standard. This conversion is normally carried out at the local telephone exchange and uses a coding method known as Pulse Code Modulation (PCM). As with all coding techniques, the PCM data stream approximates (in this case very closely) the analog input signal from the analog device, which is then converted back to an analog signal at the receiving end of the link. For all practical purposes the human ear cannot tell the difference between the original analog signal and the regenerated PCM version. Since a G.711, PCM encoded signal supports the highest level of fidelity achievable on a POTS voice line, fax machines and dial data modems have been optimized to work over these systems.

Most Carrier and Cable based VoIP solutions also encode the audio from a local handset according to the G.711 standard. This approach should, in principal at least, provide the highest quality VoIP experience and minimize the complexity and cost of broad base consumer level end-user equipment. It also makes transitions from a VoIP network to the standard TDM network relatively cost effective and straightforward.

Standard VoIP access devices therefore divide each voice call into a sequence of PCM samples of equal time period, and transmit the full 64Kbps of data within a constant stream of IP packets over the IP network. In a perfect world the resulting VoIP calls are indistinguishable in fidelity from standard POTS based telephone calls. (Cellphone calls by comparison are often compressed further so they cannot in principal be equivalent in fidelity to the best quality standard POTS calls.) Large Carriers and other mainstream service providers that control bandwidth right to the consumer can ensure that the required bandwidth is available at all times and almost achieve the goal of complete VoIP transparency compared to TDM services.

However, packet networks always incur a store and forward delay at each network node and are therefore never real time in the same sense that TDM networks are. Additionally, calls that traverse multiple carriers with VoIP segments may be subject to much variation in standard of service. A well-known VoIP experience is annoying periods of missing or garbled audio resulting from lost or delayed packets somewhere in the 'cloud'. The result is that although long distance VoIP voice calls often sound OK most of the time, (especially when compared to cell-phone calls), fax and dial modem calls frequently fail completely.

Dropped packets, latency and jitter over IP networks therefore often combine to prevent the ideal VoIP conditions from being realized in practice. In reality, fewer, smaller packets nearly always have a better chance of making it across an IP network. One technique used to improve VoIP service capability is to use IP packet header compression. More advanced products, such as NSG's, also use IP packet aggregation when possible. However, the most common process used to decrease the total amount of VoIP payload data being sent is voice compression.

4. The need for Compression

In order to best understand the effect and limitations of VoIP voice compression it is useful to initially look at the bandwidth required to transmit uncompressed VoIP calls over an IP network, and to examine the effect this has on the capacity of conventional fixed rate network links.

If we consider an uncompressed (PCM) voice call at 64Kkps and convert this to a standard VoIP call, the raw bandwidth requirement is increased to 107Kbps in each direction.

[Note: 64Kbps of audio creates 100 x 10ms voice samples per second, where each voice sample comprises 80 bytes of audio (PCM) data. (80 bytes x 8 bits x 100 samples = 64,000bits). Added to each 80 byte sample is the overhead of a TCP/IP packet which including the Ethernet frame typically comprises another 54 bytes of data. The total size of the packet transmitted every 10ms is therefore 134 Bytes, which equates to a bandwidth requirement of 107Kbps]

The impact of converting existing service platforms to standard VoIP without compression can therefore be dramatic. The raw bandwidth required to support a standard VoIP voice call is 67% greater than it is over a standard POTS network. But it gets worse. Whereas the POTS uses Time Division Multiplexing (TDM) to allocate a fixed bandwidth timeslot for each call, IP networks use contention to allocate network resource before the transmission of every packet. As a result, not only is 67% more network bandwidth required to support each VoIP call, but the contention mechanism adds even more overhead to the system. In practice any IP network loading beyond around 80% of maximum capacity significantly impedes access and congestion starts to block traffic. Overall this means that a VoIP call requires more than twice the network bandwidth of a standard G.711 POTs call to be successful.

As a result, whereas a single TDM, T1 (or Primary rate ISDN) line can accommodate 24 simultaneous G.711 voice calls, the equivalent IP bandwidth can only support 11 or 12 simultaneous VoIP calls (barely 50% of the capacity of the original TDM line) prior to the onset of failures due to congestion.

Interestingly, if the sample period of 10ms is doubled to 20ms (halving the number of packets per call), once congestion is taken into account the capacity of standard VoIP on a T1-based IP line barely increases to just 14 calls – still only 63% of the capacity of the original TDM line.

[Note: for uncompressed VoIP calls using a 20ms sample period the resulting packet size increases to 214 Bytes at 50pps, which equates to 85.6Kbps for each VoIP call.]

The basic problem is that the IP overhead is significant compared to the requirements of the packet (G.711 voice) payload. The implementation of additional services such as MPLS or VPN can make this even worse. As a result, many small Carriers and enterprises have experienced a problem when converting from TDM voice to standard VoIP calls using existing network bandwidth. Satellite networks in particular cannot cost effectively absorb the additional overhead required by VoIP and almost always use additional voice compression to alleviate this problem.

NSG VoIP products provide both standards-based and optimal (proprietary) voice compression options.

5. Voice Compression

Advanced VoIP equipment provides the option to compress the standard PCM encoded audio. The most common compression techniques use a version of a complex Algebraic Code Excited Linear Predictive Algorithm (ACELP), which requires significant Digital Signal Processing (DSP) power to implement. One such standard adopted over satellite links and some (more expensive) VoIP handsets is G.729, which reduces the audio digital data rate from 64Kbps to just 8Kbps.

Of course, any algorithm that achieves an 8:1 compression ratio is bound to reduce the fidelity of the audio signal compared to PCM. The generally accepted method of defining the audio quality over telephone networks is according to a scale known as the Mean Opinion Score (MOS). The MOS is based on subjective analysis in a controlled environment and graded between 0 and 5.0, where the perfect MOS score is 5.0. A score of 3.0 is considered slightly annoying. PCM is generally scored at MOS 4.1, which is described as perceptible but not annoying. G.729 has been a very successful standard due to the surprisingly high level of audio fidelity that is retained given the high compression ratio. G.729A is scored at MOS 3.7. Alternative versions of Linear Predictive Algorithms score as high as MOS 3.9 or more. The NSG products may be operated using both G.729 and a proprietary NSG version of ACELP, which has been independently tested in conjunction with other processes unique to NSG to exceed MOS 3.9. NSG ACELP was originally developed for military use and has been widely chosen for deployment in US Military satellite systems over the past 20 years due to its low bandwidth and very high voice quality.

The MOS of a voice call relates to the quality of the audio as it sounds to the human ear. Accordingly, the commonly used compression algorithms (including G.729 and NSG ACELP) have been tuned to minimize the audible impact of the compression on human speech. Such algorithms do not attempt to retain other components of the signal that are not critical to voice recognition, such as the specific frequency and phase relationships used by dial modems and fax machines. As a result neither of these will operate over a VoIP channel that is subjected to such compression, even though there may be minimal discernable difference in the quality of the audio to the human ear. Standard FSK tones (telephone handset key presses) can also be distorted to the point they are not decodable by the receiver after ACELP compression. The distortion is primarily introduced by the compression and regeneration of the audio signal, and may be further aggravated by the effects of network induced latency and jitter.

The benefits of compression therefore have to be weighed against its disadvantages. Furthermore, while a compression ratio of 8:1 sounds very attractive when viewed in isolation, even this is not quite as useful as it first appears, if the rest of the system is not optimized to take full advantage of the compression process.

NSG's VoIP products automatically combine compression with other processes to minimize network bandwidth requirements and maximize the quality of the voice calls.

6. Compressed VoIP Call Bandwidth

By itself, voice compression using G.729 or NSG's proprietary ACELP appear to provide a useful 8:1 reduction in the network bandwidth used, which in turn helps to improve the likelihood that VoIP packets transitioning the network are delivered successfully without interruption. However the real life bandwidth savings are not as substantial as they sound initially once the IP overhead is taken into account.

Using the same methods of calculation as above, if the audio payload of a VoIP call is compressed to 8Kbps using ACELP the equivalent calculation yields a packet size of 64 bytes, and a data rate of 51.2Kbps.

[Note: 8Kbps of compressed audio divided into 100 voice samples per second, where each voice sample comprises 10 bytes of compressed data. (10 bytes x 8 bits x 100 samples = 8,000bits). Added to each 10 byte sample is the overhead of a TCP/IP packet which typically comprises another 54 bytes of data. The total size of the packet transmitted every 10ms is therefore 64 Bytes, which equates to a bandwidth requirement of 51.2Kbps per call.]

Therefore a compression ratio of 8:1 on the voice payload yields just over 50% of bandwidth savings on the channel itself when compared with uncompressed VoIP. Often, for this reason compressed voice VoIP calls use a 20ms sample period, which adds an additional delay of at least 10ms, but yields a more acceptable 29.6Kbps bandwidth requirement for each voice call. Even so, less than 30% of the required bandwidth is being used to carry the 8Kbps voice payload.

It is clear that the IP overhead is still very significant compared to the voice payload. Additionally, each voice call is still generating 100+ packets per second (50pps in each direction for 20ms samples) so there is still the potential for a high number of packets generated by a VoIP switch or call manager to exceed the packet throughput limitations of some systems. Nevertheless independent VoIP streams using G.729 compression and a 20ms sample period is a common VoIP combination used in satellite networks. Very few products on the market are designed to improve upon this.

The solution to both these problems is IP packet shaping. NSG's VoIP products automatically use packet shaping to minimize network bandwidth requirements and maximize the quality of the voice calls.

7. Packet Shaping

Packet shaping is a term generally used to describe the optimization or control of packet services by ISP's based on specific packet types or application. Typically, the quality of service subject to packet shaping is defined in terms of jitter, latency and packet loss. Service providers use NSG products to optimize the performance of VoIP services over public (Internet) or private IP connections using a combination of voice compression and packet shaping techniques. The processes used in NSG products extend far beyond those typically used by ISPs to optimize VoIP services and are based on the proprietary NSG trunking protocol known an SFTM (Switched Frame Transfer Mode).

SFTM is a proprietary networking protocol used between NSG's VoIP products to optimize trunk bandwidth, Quality of Service (QoS), and call routing. SFTM is more efficient than standard IP based protocols (such as the Real-time Transport Protocol - RTP) for several reasons. Firstly, SFTM uses Basic IP datagrams (UDP packets) rather than full IP packets whenever possible. Secondly, the SFTM header information itself is shorter than either standard RTP or compressed RTP. Thirdly and most significantly, multiple packets to the same destination are automatically multiplexed into a single packet to dramatically reduce the total number of packets that need to be transmitted between nodes, and as a result reduce the bandwidth used overall.

For example, using these NSG techniques, even a single compressed VoIP call over an IP network can be reduced from 29.6Kbps (see above) to just ***15.2 Kbps*** per call. When just three calls are active the bandwidth drops to only ***11 Kbps*** per call. When ten calls are active the bandwidth used is just ***9.5 Kbps*** per call. The fidelity of the audio is unchanged. In fact the overall quality of the call is normally improved compared with standard VoIP, since fewer, smaller packets are less likely to be delayed or dropped by the network.

Such dramatic bandwidth reduction is a remarkable, industry-leading result that many potential customers do not believe until they see it demonstrated in their own lab. This has been independently tested and verified by iDirect in the USA and others internationally.

The practical effect of such a dramatic bandwidth saving cannot be over emphasized. For example as discussed above, a T1 rate TDM line than supports 24 POTS calls, when converted to IP only supports somewhere in the range 11 – 14 standard (uncompressed or compressed) VoIP calls. **However, using NSG SFTM technology the same T1 bandwidth can comfortably support *over 120* toll quality VoIP calls.**

Furthermore the total number of packets required to support 100 simultaneous voice calls can be as few as 100pps in each direction (depending on configuration setting). This compares with 6,000pps in each direction for standard compressed VoIP (20ms sample period = 50pps x 120 calls = 6,000pps). As a practical configuration, when a large number of calls are being supported the NSG recommended setting is in the region of 300 – 500 pps in each direction. This minimizes latency and keeps packet sizes to within reasonable limits.

It should be noted that these bandwidth and packet throughput savings are not obtained by using more aggressive methods such as silence suppression or ultra low bandwidth voice compression. The voice quality of these calls using NSG ACELP can easily exceed a MOS of 3.9 and are indistinguishable by most people from conventions POTS telephone calls. G.729 compression can also be used, and may be advantageous if the requirement is to interface directly to external VoIP network equipment based on this standard.

Of course, the beneficial effects of SFTM can only be realized directly between NSG VoIP switching nodes or access devices, and any links that interface with outside equipment must conform to external standards. **For this reason NSG VoIP products use advanced packet shaping capabilities between nodes and also provide gateway capabilities to ensure interoperability with standard VoIP equipment, including many VoIP handsets, SIP servers, soft-switches and call managers.**

8. <u>Coping with Latency and Jitter – Voice Calls</u>

Regardless of the compression technique used, latency and jitter are inevitable when using VoIP connections. In very fast private networks and when used by Carriers for local access, latency and jitter may be minimal and the effects inconsequential, but over public IP or Internet connections they vary greatly and can be significant. Satellite based systems of course cannot avoid a 250ms trip delay although they may exhibit minimal jitter. 3G cellular data wireless connections often exhibit delays measured in many hundreds of milliseconds, sometimes up to a second, and jitter that is not much less. 4G cellular data is normally far less prone to jitter or delay than 3G network data but both are still present.

The primary way to accommodate jitter is to use a jitter buffer that exceeds the maximum expected variation in delay over the network. However, if the delay is maintained at much more than a few hundred milliseconds it becomes very noticeable and eventually unacceptable, even for voice calls. Some early VoIP systems used a variable delay where the jitter buffer effectively expanded to accommodate the maximum delay on a call so far. Unfortunately this resulted in the delay gradually growing during the period of a call, often to a point where the connection was no longer comfortable to use, at which time the only choice was to hang up and redial. The best VoIP systems, such as NSG's, now keep the delay constant and cover for missing or late packets using unobtrusive methods.

The primary technique already mentioned is to reduce the likelihood of missing packets by keeping the packets small using voice compression and packet shaping. From the users perspective the objective is to cover for missing voice samples using a technique that joins separated voice samples together without the listener being aware of any discontinuity or strange clicks or jumps. Sometimes re-processed voice samples may be substituted for lost packets in order to keep the audio flowing normally and provided there are only isolated packet losses the user is not normally aware of such refinements.

NSG VoIP products use a variety of advanced techniques for coping with missing voice samples to minimize the audible impact to users.

9. Coping with Latency and Jitter – Fax/Modem Calls

Where real-time fax and modems calls are concerned, lost packet data can easily create a terminal problem at the receiving end of the link. Special handling is required for each type of transmission. NSG Patented Artificial Intelligence (AI) techniques used in ***Netrix*** brand VoIP equipment include the capability to identify certain communication sequences that might indicate a fax, modem or other type of tone based 'data' call is taking place and to operate in a different mode from that point in the call onwards. The basic processes involve a combination of techniques that include tone detection, digital transmission of action events, decoding standard modem data and the ability to detect and absorb repeated messages that occur due to timeouts and/or lost packets during non-critical portions of the communication. There are a number of different modes that can be used depending on the application and modem protocols being used.

Real time fax support in NSG products uses some of the techniques employed by the T.38 standard, which are enhanced by Patented NSG techniques to accommodate the effect of extended jitter and delay. As a result, typical G.711 or T.38 fax calls that fail over a VoIP network can be successfully made using the NSG VoIP equipment. This includes the reliable support of fax calls over 3G cellular wireless data networks that might incur delays and jitter of up to 1 second or more. **The NSG *Wi-Modem* has been successfully tested and approved by Sprint and Verizon Wireless as able to successfully transmit faxes over the 3G cellular data network.**

Dial modem calls and other FSK tone based communications (such as alarm panels or simple touch pad key presses) can also be supported in real time rather than using the more frequent and less desirable technique of store and forward, also known as modem relay. Another fairly simple and widely employed technique is to recognize and regenerate tone sequences at the far end of the link, which avoids distortion due to compression or lost packets. This latter capability is enhanced with NSG's AI software that manipulates certain parameters in the transmission and response sequence in order to improve dependability over IP data networks.

A combination of the above techniques can be utilized to optimize NSG VoIP products for use in new applications.

10. Advanced Call Connect Capabilities

There are a number of applications for which conventional VoIP solutions are not ideal due to the way VoIP calls are connected using the Session Initiation Protocol (SIP). SIP calls often depend on the use of a dedicated server or centralized call manager to provide call connect information. One of the main criticisms levied against SIP is that it is a very bandwidth intensive protocol. SIP messages have been designed to be human readable and contain copious plaintext. The protocol itself follows a request/response format and each SIP message can extend to hundreds of characters (bytes) in length. While this is not a big issue for high-speed networks, on satellite and wireless links the additional overhead can be costly and the delay in establishing call setup very noticeable.

In addition to supporting standards based SIP call connections, NSG products also support a much more efficient pier-to-pier connection protocol within its SFTM trunking scheme that uses minimal bandwidth for call setup and clear. **SFTM provides fast connection times (much faster than standard VoIP using SIP) and saves bandwidth on expensive links.**

Using SFTM it is also possible to leave in place a constant connection between network nodes utilizing minimal keep-alive traffic. For Satellite links the keep-alive can be turned down or even turned off completely even though a VoIP call over the connection may still be activated immediately without any further link setup required. If the trunk is disturbed for any reason and does need to be reestablished, this can be achieved within a few seconds of a call being initiated. SFTM trunking with minimal keep-alive may be used on both point-to-point (SCPC) and fully meshed (DAMA) satellite systems.

If a primary site is down or busy, SFTM provides automatic re-routing to a backup or secondary site. Similarly, SFTM supports least cost routing and automatic rerouting in the case of a network link failure.

SFTM also provides a capability equivalent to a 'hoot and holler' connection between handsets that is considered essential for some applications such as in brokerage houses or for emergency services. Conventional VoIP systems cannot support immediate connection between handsets and for this reason VoIP is often not considered as a viable option for these applications. However NSG VoIP products are currently being used by customers to provide this capability.

In conjunction with hardware options available on some NSG VoIP products, SFTM also supports an automatic ring-down feature. This capability can be used to control signal lines to external equipment such as Radio Transmitters and is a particularly useful when used in Air Traffic Control systems.

11. VoIP Trunk Optimization

The capabilities of NSG VoIP products to improve the efficiency of VoIP services over connections between NSG nodes can also be used to improve capacity of standard VoIP calls over trunks with limited bandwidth. An obvious example is in satellite networks, but the throughput of wireless and terrestrial connections can also be greatly improved using VoIPAK and VoIPZIP functions.

VoIPAK and VoIPZIP both use the SFTM trunking as described above to optimize generic VoIP services. Standard VoIP calls made between conventional VoIP equipment (manufactured by other companies) can be optimized by eliminating packet overhead and combining multiple calls into a single packet stream. Additionally, VoIPZIP provides voice compression of standard G.711 based VoIP using either NSG ACELP or G.729. VoIPAK and VoIPZIP both support point-to-point and fully meshed connectivity.

VoIPAK is therefore normally used to further improve the efficiency of VoIP services when the payload is already compressed using (typically) G.729. VoIPAK is often used on satellite links that already employ voice compression technology and where SFTM packet shaping can provide additional bandwidth optimization of up to 3:1.

VoIPZIP is often used to compress standard VoIP calls on wireless and. terrestrial links. The compression algorithm of choice is NSG ACELP, which combined with packet shaping can result in an overall optimization advantage of 10:1 or more.

12. Silence Suppression

In addition to packet shaping, a further optimization technique available on NSG VoIP products is silence suppression. Silence suppression is a method whereby the (compressed or uncompressed) audio during periods of silence is not transmitted over the link. Instead, short commands are sent between nodes to indicate the period of silence, and the output audio at the receiving and of the link is stopped during that period. An audio level energy threshold is set to trigger periods of silence suppression on and off at the transmitting end of the link.

Depending how it is implemented, silence suppression can either be unobtrusive or be very noticeable to users with the potential for abrupt cuts in the audio at the beginning and end of speech segments - often described as clipping. As a result silence suppression is generally avoided and only used when bandwidth is at a premium.

The effectiveness (and intrusiveness) of the silence suppression mostly depends at what level the threshold is set. If silence suppression is set very aggressively the audio payload bandwidth savings can be up to 50% (since theoretically only one person is talking at a time). However most people find this level of suppression very intrusive. More realistic savings are closer to 30% when the audio threshold is lowered to a less intrusive level, although it is still noticeable to most people.

One of the capabilities of NSG VoIP products is to add in “comfort noise” during periods of silence when silence suppression is enabled. Comfort noise level can be set to mimic the background channel noise normally heard when there is no audio activity, effectively making the periods of active silence suppression less noticeable to the listener.

The ability of NSG VoIP products to combine comfort noise with silence suppression can be used to advantage on bandwidth-restricted links as an alternative to implementing more aggressive voice compression. This can yield an overall optimization advantage of up to 16:1 while still retaining PSTN quality audio with minimal clipping.

13. Gateway to the PSTN

VoIP calls are made over IP data networks and can only connect directly to IP based servers or other directly connected VoIP devices or equipment. In order to allow general access to the PSTN and the ability to call (for example) home phones and cell phones, any VoIP call originating from a public or private IP network needs to pass through a media gateway device that translates the call from a VoIP call to a standard PSTN call. This gateway function can also provide the VoIP user with a PSTN Direct Inward Dial (DID) telephone number, which facilitates incoming calls from the PSTN.

Carriers and Cable companies offering generic VoIP telephone service provide a transparent pathway to the PSTN through Softswitches. 'Free' VoIP service providers such as Vonage and Skype also provide PSTN gateway connection to their customers for a fee.

NSG VoIP products support gateway capabilities that allow connection between our proprietary SFTM trunking, the PSTN and standard SIP based VoIP networks. There are a number of ways this connection can be accomplished.

The simplest way to achieve this is to connect a NSG VoIP gateway (T1/E1 or Analogue) to an existing private PBX or Key system. Each telephone handsets (or other device) that connects over the VoIP network to the NSG gateway is allocated a unique extension number on the PBX. Using this configuration each connected analog device appears to be part of the PBX system and there is no difference in operation between these remotely connected VoIP handsets and handsets directly connected to the PBX. When a VoIP user lifts the handset the dial tone comes directly from the PBX. Incoming calls from the PSTN may be received directly and outgoing calls may be made dialing the required PBX sequence (normally 8 or 9) to connect to an outside line. The VoIP users can be spread throughout the world and all have local PBX telephone numbers, appearing from the outside to all be located in the same place.

An alternative way to connect private NSG VoIP calls to the PSN is through a Competitive Local Exchange Carrier (CLEC) that can provide PSTN access through an existing NSG gateway. The local Carrier provides a Direct Inward Dial (DID) telephone number that is unique to the VoIP user and allows incoming and outgoing PSTN calls through the NSG gateway. Direct VoIP calls between IP devices do not need to pass through the Gateway of course.

A third alternative which is a variation of the above, is for an intermediary party to provide a managed VoIP service to its clients using its own NSG gateway for PBX or PSTN access. In this case the intermediary service provider would be NSG's customer.

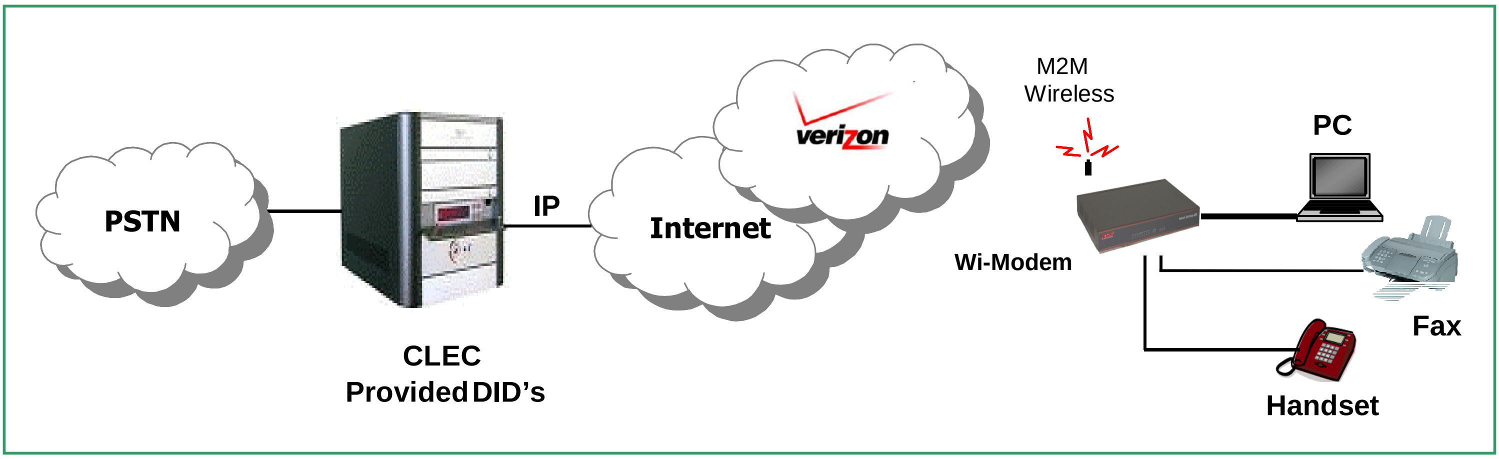
There are many ways in which NSG VoIP equipment may be utilized for primary and/or backup access into private and public telephone networks. Typical customer applications for NSG VoIP products are outlined below.

14. Some Key applications

Mobile office: Using the 3G wireless service to support 2 voice/fax lines and Internet Access over a single wireless connection. The voice lines can be extensions off an HQ PBX or be direct to the PSTN through a gateway.

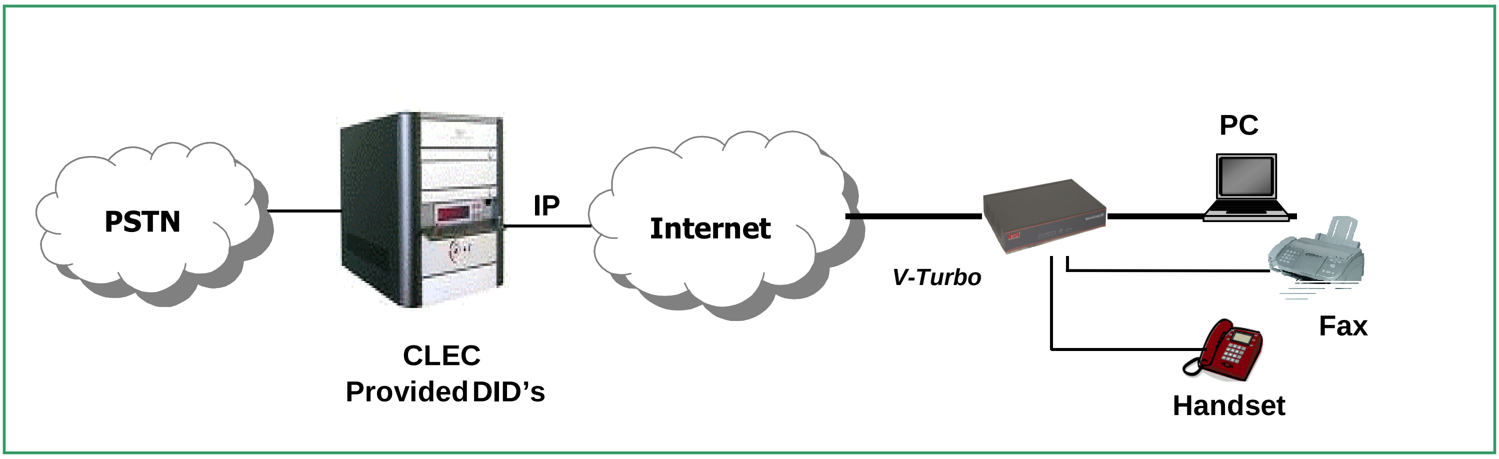
Key advantages: Mobility,
PSTN quality voice (not cell pone quality),
Fax Support
Internet (PC) access

Multiple services are provided over a single, cost effective 3G wireless connection



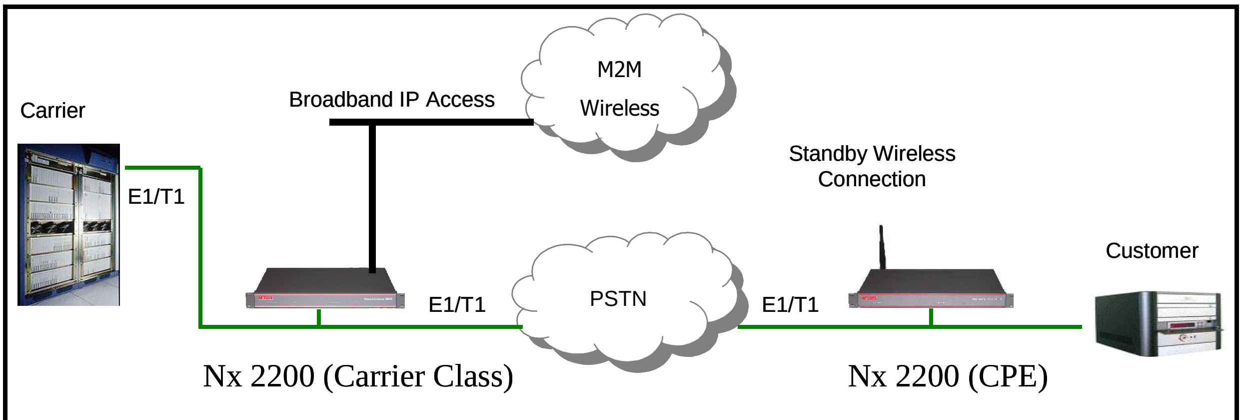
Remote office: Using direct Internet (eg. Cable) IP access provides 2+ voice/fax lines and/or Point of Sale terminals. The voice lines can be extensions off an HQ PBX or be direct to the PSTN through a gateway. POS terminals may directly access IP hosts.

Key advantages: PSTN quality voice (not cell pone quality),
Fax Support
POS (Modem) support (possibly direct into IP Host)

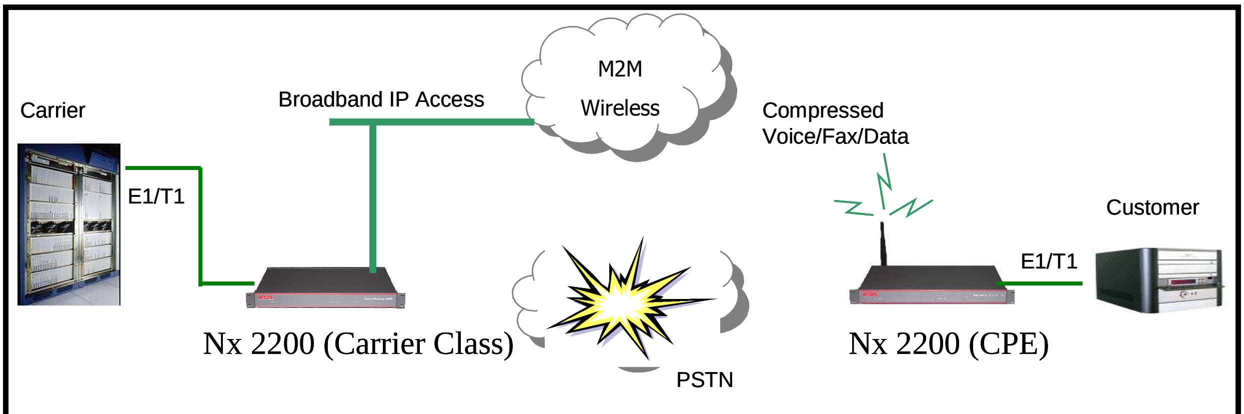


Automated backup to primary voice service: One of the options available on NSG devices is the ability to monitor an existing TDM PSTN digital trunk and provide an automatic backup to that service over and IP network using VoIP over wireless or standard Internet connections. This capability can also be used for overflow traffic during emergency or peak periods.

Key advantages: Automatic Monitoring and Backup of critical TDM Services
PSTN quality voice (not cell pone quality),



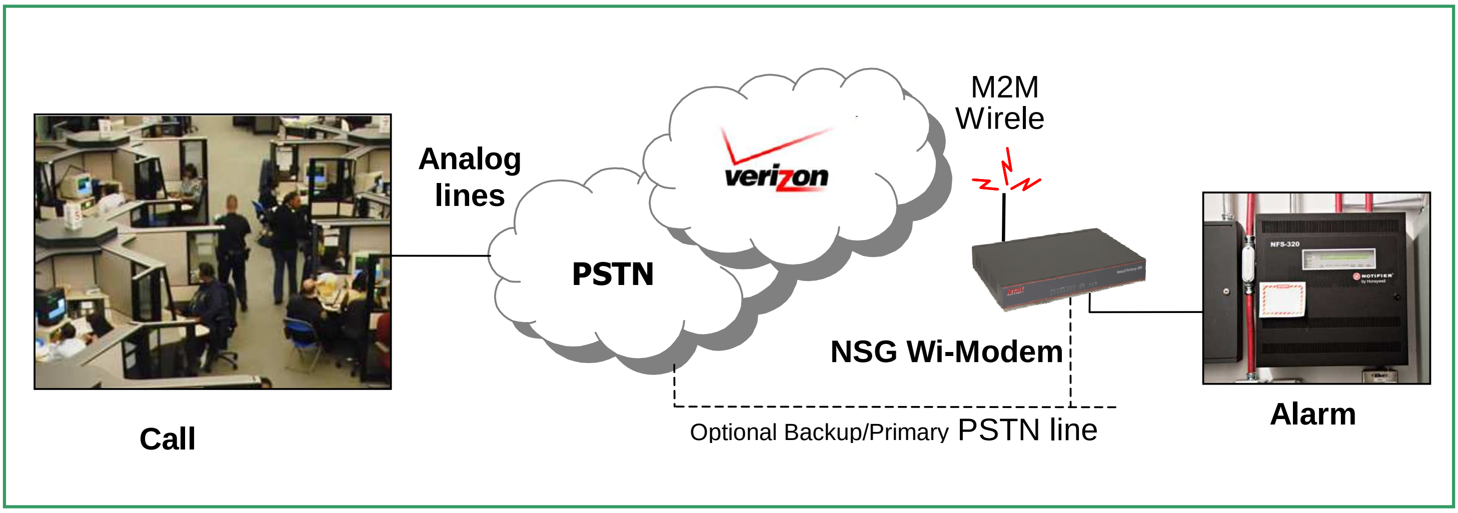
Normal Operation - Units in Unobtrusive Monitor mode



Emergency Operation - Units Switch into Circuit and Connect using Wireless M2M network

Dial Modem Support: Using 3G or direct Internet (eg. Cable) connection provides access to devices with an installed modem such as Alarm Panels, Telemedicine Machines, SCADA devices etc.

Key advantages: Real time Modem support over IP connections
PSTN quality voice channel also available
Serial data port also available



Hoot and Holler: Provides Support for immediate call connections such as used in Brokerage house applications.

Key advantages: Immediate voice connection over IP
Retains PSTN quality voice

Equipment Control: Provides passing of external control signals (through E&M signaling) for the control of external equipment such as keying Wireless transmitters.

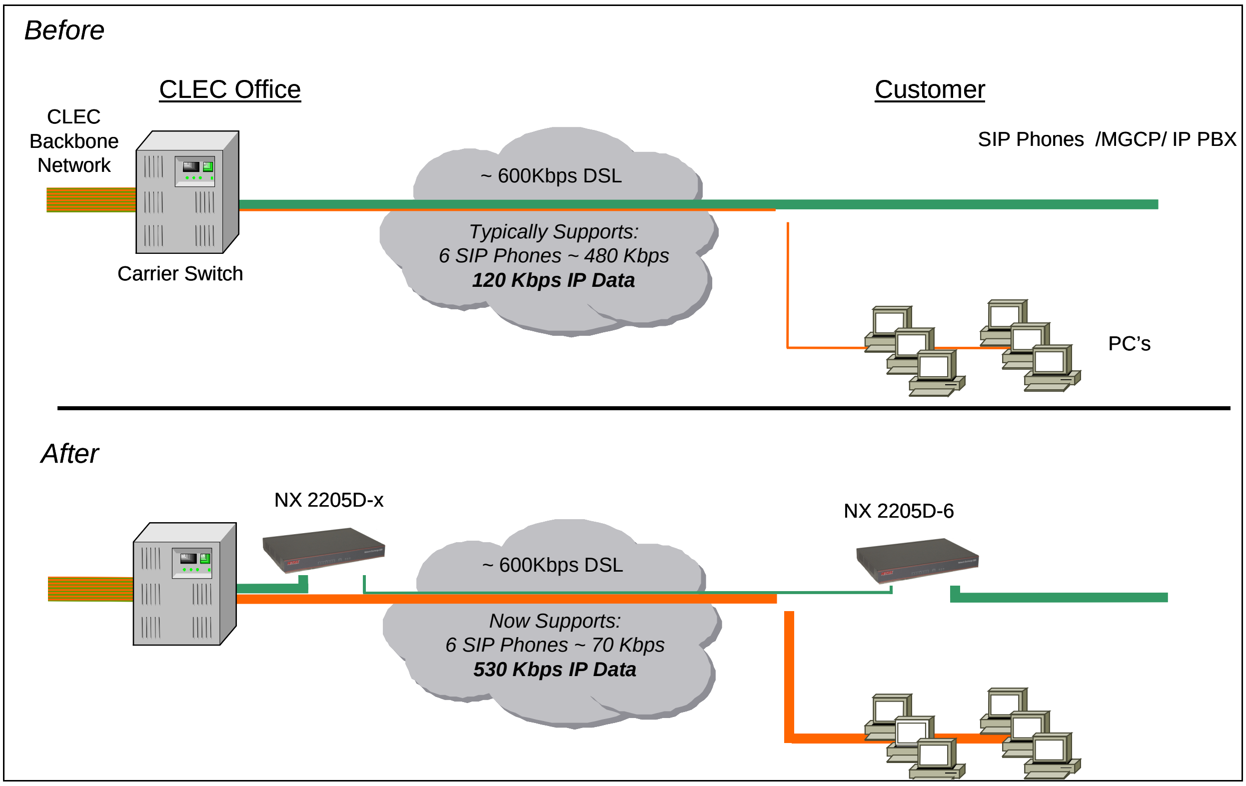
Key advantages: Immediate connect over IP
Retains PSTN quality voice channel

Satellite Networks: Cost reduction and Capacity improvements over fixed SCPC and IP based DAMA satellite links.

Key advantages: Dramatic reduction in VoIP Bandwidth requirements
Speeds up connect times
Reduces Packet throughput
Large cost savings for operators

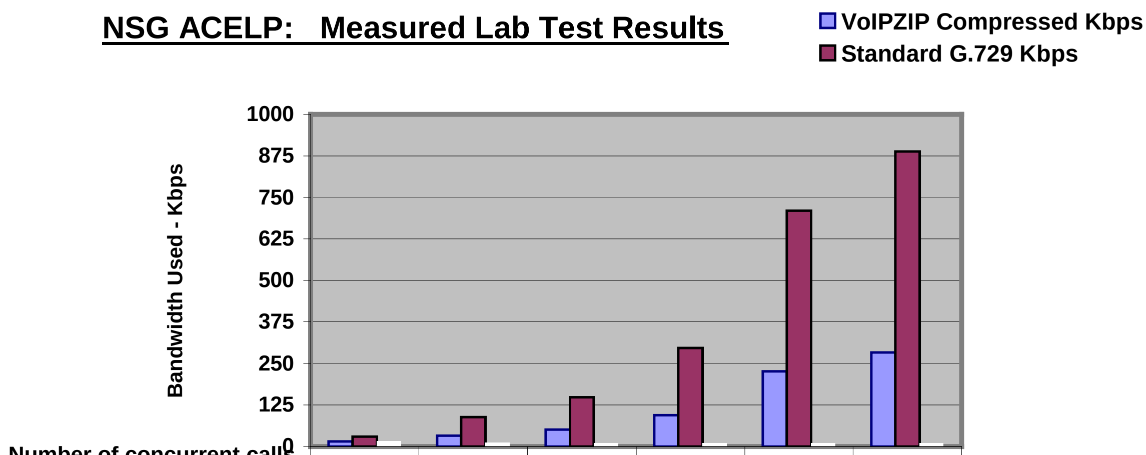
VoIP Trunking: Improves throughput capacity on terrestrial VoIP trunks.

Key advantages: Dramatic reduction in VoIP Bandwidth requirements
Retains PSTN quality voice
Large cost savings for operators



15. NSG ACELP Performance graph

The graph below shows typical lab results comparing the bandwidth advantage of using NSG ACELP (18ms sample periods) with standard compressed G.729 VoIP (20ms sample periods) over VoIP trunks.



Number of concurrent calls	1	3	5	10	24	30
VoIPZIP Compressed Kbps	15.2	33	51	95	226	283
Standard G.729 Kbps	30	88	148	296	710	888
Bandwidth per call	15.2	11.0	10.2	9.5	9.4	9.4

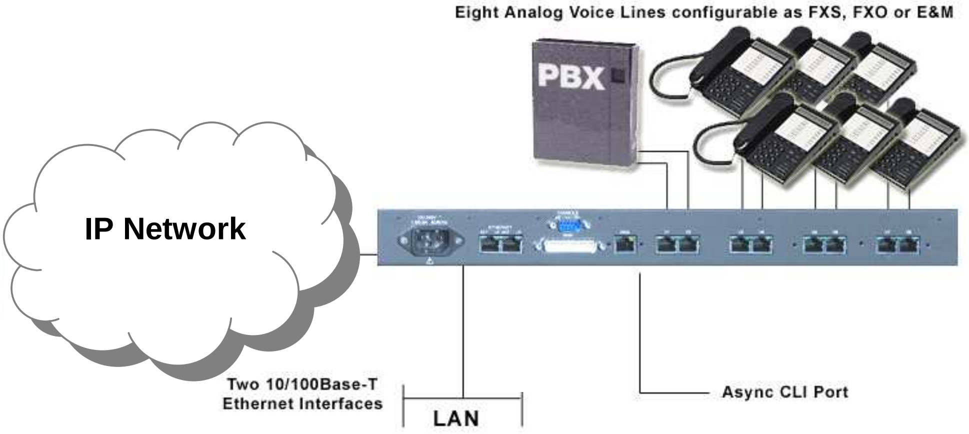
16. <u>Products</u>

V-Turbo: Access device with 2x analog voice/fax/modem ports, 1x Ethernet and 1x Serial port.

Wi-Modem: 3G Wireless access device with 2x analog voice/fax/modem ports, one Ethernet and one Serial port (optional external USB rather than internal Wireless module).

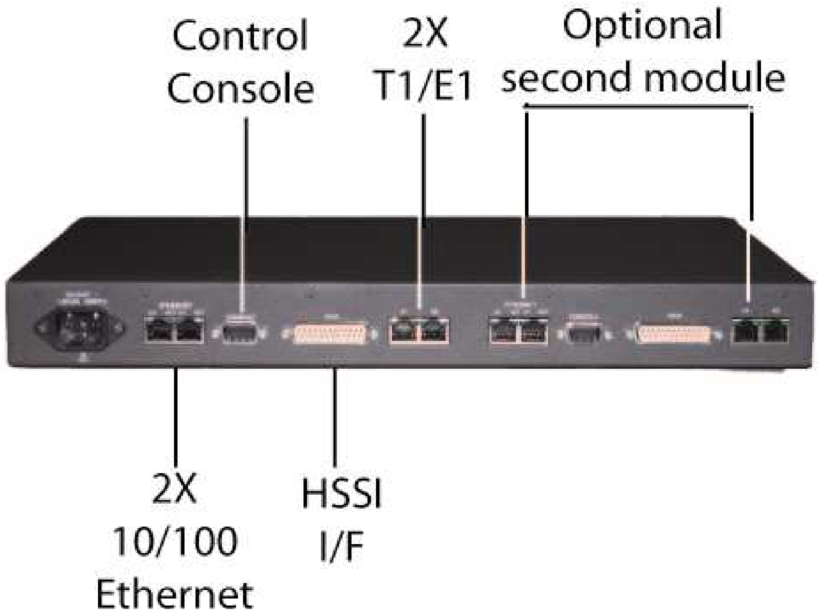
2205A/2/4/8 Analog Access gateway: 2 to 8 analog voice/fax ports, 2x Ethernet ports and 1x Serial Port.

2205A/2: 2 Analog port unit



2205A 4/8 Analog port unit (19" rack mount)

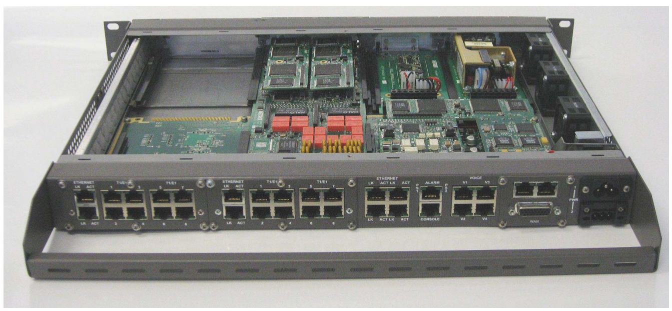
2205D Digital Access gateway: 1x T1/E1 Digital voice/Fax port, 1x T1/E1 data port, 2x Ethernet ports and 1x Serial port.

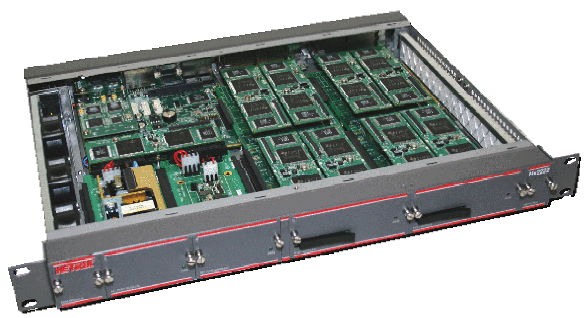


VoIPAK IP Packet shaping: Software or Hardware platform for optimizing VoIP packet trunks by packet shaping. Based on 2205D.

VoIPZIP IP Packet compression and shaping: Hardware platform based on Nx2205D comprising 1x T1/E1 Digital voice/Fax port, 1x T1/E1 data port, 2x Ethernet ports and 1x Serial port for optimizing VoIP packet trunks using DSP based voice compression and packet shaping.

Nx2222 VoIP Services Aggregation and Concentration Gateway: Modular Central Office concentration and/or remote access mediation device supporting up to 500 voice channels on a combination of digital Trunks (up to 18) and Analog ports (up to 28) high speed data serial links (up to 10) and Ethernet ports (up to 6).





17. Bullet Summary

- NSG's ***Netrix*** brand VoIP Products offer greater throughput, higher voice quality and more capability than conventional (standard) VoIP equipment.
- NSG VoIP products work over connections where other VoIP equipment repeatedly fails, eg. 3G Wireless.
- Standard VoIP calls require much more bandwidth to support than standard PSTN calls, (typically around 100Kbps compared to 64Kbps).
- Standard VoIP compression helps but does not alleviate all the problems when bandwidth is expensive or packet throughput is limited.
- NSG products address both these problems using its SFTM protocol, firstly by compressing the Voice and secondly by combining and streamlining packets (packet shaping).
- The resulting bandwidth occupied by a single NSG VoIP call ***including the IP and Ethernet packet overhead*** can be less than 9.5Kbps.
- Even at 9.5Kbps, NSG VoIP products maintain high quality voice indistinguishable from the PSTN, with a MOS of 3.9.
- NSG Products also provide additional functionality using a combination of Proprietary and NSG Patented techniques.
- Such techniques provide support for enhanced voice services, and support faxes and dial modems that extend beyond the normal capabilities of VoIP.
- Verizon and Sprint have both certified operation of NSG products over 3G Wireless.
- NSG Products provide gateway functions to standard VoIP services and to the PSTN.
- These are some of the reasons that the US Military as been using NSG VoIP solutions for the past 20 years.

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