LAN/WAN GUIDELINES

VoIP (Voice over Internet Protocol) is simply the transmission of voice traffic over IP-based networks. It is not voice over the Internet. Voice over the Internet would be Vontage or Skype.

VOIP gives you the ability to use your existing LAN/WAN infrastructure that you already own and allows your voice traffic to pass over it. When you utilize VOIP on your data network it works exactly like desktop computer system does on your network, it sends packets of information out and the data network analyzes these packets and determines where they should go and how they should get there based on the attached IP address header. When you use a VOIP telephone system it is connected to your data network and as you speak over the phone system it converts your conversation into packets, much like your work station does, but these packets are called voice packets. With a VOIP system it is critically important that your existing data network be able to decipher between voice and data packets. The reason this is so critical is because when you are doing basic data computing over your network it is not all that critical that all those data packets arrive at the same time. If there is a slight delay of the transmission of the data packets it is transparent to you, as a user you wouldn't ever even notice it. But if there is any delay what's so ever with a voice packet, you will notice it immediately by experiencing what's known as delay or jitter. This would be very similar to what you get on a bad cellular call when you experience very choppy and delayed voice transmission. In order to prevent this we need to ensure that your existing data network will support voice over it.

Here are the simplified guidelines required to support voice packets over the data network:

QOS – QUALITY OF SERVICE (LAN SIDE) *– This is a programmable option in both data switches and routers which allows the network to differentiate a voice packet from a data packet. By identifying the voice and data packets independently, you must always give the voice packet a higher priority which should prevent the VOIP call from experiencing any delay or jitter problems which would create a poor quality call.* ***All hubs must be replaced by programmable data switches.***

QOS – QUALITY OF SERVICE (WAN SIDE) *– This can be accomplished in two independent ways. The first way is through programming options on the routers. The second method is through the carrier. There are some carrier circuits out there that provide QOS over the WAN, such as an MPLS circuit.*

POWER *– VOIP phones each require their own power source. This can be done in a variety of methods. It can be done through the data switch through a methodology called POE (power over the Ethernet), basically the data switch port provides the power to the VOIP phone and is there for the phone when you plug it into the wall jack. Another methodology of proving a VOIP phone power is through the use of a power injector. This methodology is done by placing the power injector unit in line with the data switch port which would then supply the VOIP set its power. To the end user this would look identical to a POE data switch. The last methodology is*

to plug a power brick or transformer into the VOIP set. This is cumbersome because each VOIP set must be in a reasonable proximity to a commercial power outlet.

BANDWIDTH *– An uncompressed VOIP call (G.711 codec) uses about 100 kbytes of bandwidth, a compressed VOIP call (G.729 codec) uses about 33 kbytes of bandwidth. This consideration is far more applicable over the WAN that it would be over the LAN. This is because bandwidth on the LAN is abundant and basically free, but bandwidth over the WAN is a commodity that the end user pays for.*

CABLE *– A minimum of category 5E cable, terminated on RJ45's at both ends, and wired as an industry standard 568B configuration is required. Category 3 will work, but only up to 10baseT which is considered an archaic transmission by today's standards and highly not recommendable.*

IP ADDRESSING FOR VOIP TELEPHONES *– When a VOIP device is sitting on the data network it needs to communicate with the data network so the data network can accept and pass the voice packets the VOIP telephone is sending. In order to do this each VOIP device needs its own independent IP address on the data network. There are two ways that this can happen. The industry standard method is to distribute IP addresses through DHCP (Dynamic Host Configuration Protocol). DHCP can either be delivered using a customer provided DHCP server, or the IP PBX can also deliver them to the IP devices. The other method of delivering IP addresses is to assign and hard program each device with its own independent static IP address. This practice is generally not used because it is very time consuming and inflexible.*

SUBNETS *– Generally this is a good thing to do, irrespective of whether a VOIP installation is being used or not. Without the use of subnets you have a flat network. Creating a flat network may appear to speed up transactions because there are less links to pass through, but in reality this is not the case. A flat network can potentially grind to a halt, not because of genuine traffic, but simply due to the large amount of broadcast traffic required to reach all devices. Using subnets along with layer 2/3 data switches helps illuminate this by segmenting broadcast domains.*

VLAN – VIRTUAL LOCAL AREA NETWORK *– VLAN's should be used in conjunction with subnets. It is a method of segregating the voice packets from the data packets with a methodology called VLAN tagging by dividing what is one physical network into multiple virtual networks.*

In conclusion when assessing your network if you can adhere to the guidelines above and specifications below, you will have very high quality, if not perfect voice transmissions:

Packet Loss	Jitter	End-to-End Delay	Ping' Delay	
<1%	<30ms	<50ms	<100ms	Green - Go!
<5%	<60ms	<80ms	<160ms	Amber - Caution
>5%	>60ms	>80ms	>160ms	Red - STOP!