GUIDELINES FOR VOIP NETWORK PREREQUISITES

WHITE PAPER

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MATRIX
TELECOM SOLUTIONS

Executive Summary

This document contains basic network requirements that are foundational for good voice quality when using MATRIX UC Server/IP-PBX and GATEWAY products solutions over a data network. No document can satisfy detailed needs of every network and therefore this paper serves only as a starting point. The document summary provides a short list of networking requirements, allowances and recommendations. Use this document as a checklist to determine if the network meets the minimum requirements for implementing Voice over Internet Protocol (VOIP) with acceptable quality. It also explains why VOIP applications can yield poor results when data traffic on the same network doesn't seem to have problems.

Voice quality is always a subjective topic. Defining good voice quality varies with business needs, cultural differences, customer expectations, etc. The requirements below are based on the ITU-T, EIA/TIA guidelines. Note that while Matrix's requirements will meet or exceed most customer quality expectations, the final determination of acceptable voice quality lies with the customer's definition of quality and the design, implementation and monitoring of the end to end data network.

Quality is not measured by one discrete value where a number 8 is good and 9 is bad. There is a tradeoff between real-world limits and acceptable voice quality. Less delay, jitter and packet loss values can produce the best voice quality, but may also come with a cost to upgrade the network infrastructure to get to the lower network values. Another real world limit is the inherent WAN delay over a trunk linking, for example, the U.S. West coast to India. This link could easily add a fixed 150ms delay into the overall delay budget and is beyond the control of an enterprise. Perfectly acceptable voice quality is attainable but will not have toll quality.

VOIP NETWORK CHECKLIST PARAMETERS

Network Delay/Network Latency

Network delay is an important design and performance characteristic of a computer network or telecommunications network. The delay of a network specifies how long it takes for a bit of data to travel across the network from one node or endpoint to another.

Callers usually notice round trip voice delays of 250ms or more. ITU-T G.114 recommends a maximum of 150ms one-way latency.

Since this includes the entire voice path, part of which may be on the public Internet, your own network should have transit latencies of considerably less than 150ms.

Network Jitter

Jitter is a common problem of the connectionless networks or packet switched networks. Because the information (voice packets) is divided into packets, each packet can travel by a different path from the sender to the receiver. When packets arrive at their intended destination in a different order then they were originally sent, the result is a call with poor or scrambled audio.

Jitter is technically the measure of the variability over time of latency across a network. Jitter is one of the most common VOIP call quality problems.

Toll quality suggests average jitter be less than ½ the packet payload. This value has some latitude depending on the type of service the jitter buffer has in comparison to other buffers, packet size used, etc.

Jitter that exceeds 40ms will cause severe deterioration in call quality. The solution of this problem is to use Jitter Buffers. A jitter buffer temporarily stores arriving packets in order to minimize delay variations.

Matrix VOIP interface supports Jitter Buffer settings to solve Jitter problem. Static or Dynamic Jitter buffer value with minimum delay (ms) and Optimization factor can be set.

Network Packet Loss

Network Packet loss is the maximum loss of packets (or frames) between endpoints.

VOIP is not tolerant of packet loss. Even one percent packet loss can "significantly degrade" a VOIP call using a G.711 codec and other more compressing codecs can tolerate even less packet loss.

The default G.729 codec requires packet loss far less than one percent to avoid audible errors. Ideally, there should be no packet loss for VOIP.

Connection Speed and Codecs

VOIP transmits voice data packets in a compressed form, so that the load to be transmitted is lighter. The compression software used for this are called codecs.

Codec's main function is Encoding-Decoding, Compression-Decompression, and Encryption-Decryption.

Matrix provides following codec support for IP calling. G.723 (6.3 kbps and 5.3 kbps), G.729 AB (8kbps), GSM FR, ilbc-30 (13.3 kbps), ilbc-20 (15.2 kbps) ,GSM FR (6.5 to 13 kbps), G.711 A/U law (64kbps).

Codec values and bandwidth might vary from manufacturer to manufacturer as each has different overhead values and different implementation logic.

Connection speed (technically called the bit rate) is measured in Kilobits per second (Kbps). It is simply a measure of how many bits are transmitted in one second. Every service provider talks about connection speed when referring to the speed they offer.

If you have dial-up connection, do not expect great quality. A broadband connection will work right, as long as it is not spotty, and not shared with too many other communication applications.

Below are some typical bandwidth values associated with popular communication devices and technologies

Technology	Speed	Use in VOIP
Dial up (Modem)	Up to 56 Kbps	Not Recommended
ADSL	Up to several Mbps	Recommended
LAN (e.g. Ethernet)	Up to thousands of Mbps	Highly Recommended for best performance
Cable	1 to 6 Mbps	Recommended

Bandwidth Requirement for VOIP Calls

Bandwidth refers to the data transfer rate of your Internet service. It describes the amount of data that you can transfer over your Internet service during a specific time. The bandwidth you receive from your Internet Service Provider (ISP) is important because you need to be able to allocate a certain amount to your VOIP service.

VOIP Phone systems run completely over the Internet. Because of that, your call quality is 100% dependent on the Internet service that you use. Low bandwidth can lead to poor call quality due to slowly delivered data packets, or even a system that cannot send or receive calls.

You may not need any additional bandwidth to support high quality VOIP calls but you will have to evaluate your current Internet service in order to find out.

Required bandwidth for business quality communication depends on the Average number of simultaneous call and Codec used.
Bandwidth Calculator will help you define the required bandwidth based on the codec you will choose and the number of simultaneous calls required for communication.

Please find below link to download Bandwidth calculator.
http://www.matrixtelesol.com/Bandwidth-Calculator.xlsx

Video calls with H.264 codec may use typical bandwidth of 800 Kbps with 640x480 resolution and 4:3 aspect ratio. Other video codec may use different bandwidth according to resolution and aspect ratio.

Pre-requisites when UC Server/IP-PBX and IP Phones Configured under VLAN Environment

It is highly recommended to create separate VLAN for voice and data networks, when IP Phones and PCs are connected in same switch.

Manageable L2/L3 Switch is highly recommended for such kind of network setup.

Required throughput or data capacity of switch should be calculated properly, based on device connected to switch and its bandwidth requirement (i.e. 1Mbps, 10Mbps, 100Mbps or 1Gbps per port).

For better voice quality it is advisable to configure CoS (Class of Service) in L2 Switch and QoS (Quality of Service) in Router, which will provide priority to voice packets in buffer.

Matrix recommends below value for QOS and CoS for optimum Voice and Video performance.

Application	Layer 3 (QoS)	Layer 2 (CoS)
VoiceInteractive	DSCP-46	5
Video Interactive	DSCP-34	4
Streaming Video	DSCP-32	4

Pre-requisites when UC Server/IP-PBX and IP Phones Installed under VPN Environment

UC Server/IP-PBX may also work in the VPN environment created between multiple locations. VPN tunnel can be created from VPN protocol and VPN network can be set.

For successful VOIP calling between multiple locations, seamless connectivity must be achieved between each location.

There should not be ping drop or more latency between locations. This might result in delay in voice packets or echo in speech.

VPN tunnel configuration has settings for "Tunnel Keep alive" and "Ideal time to disable VPN tunnel when no data is received". These settings shall be configured for optimum performance from VPN.

Static IP Requirement for External Communication

Static IP is required when multiple locations are interconnected over IP or IP Phones to be used from external network.

When Matrix UC clients in Mobile are required to use from external 3G/4G network Static IP is still required at the UC Server/IP-PBX location.

There are scenarios where, static IP is not present and internet is available from any dynamic source. In this condition Dyn DNS and STUN is required to configure for accessing PBX from external network.

It is noticed that where multiple static IP are available and Load Balancing is used in Router/Firewall, Peer-to-Peer calling may not work seamlessly as IP changes gradually. Additionally, it is noticed that traffic handled by Router/Firewall is random in manner, which means outbound and inbound traffic is handled via different WAN interface and in this case SIP will not work.

The Matrix system sends outgoing peer-to-peer traffic only on one Static IP and it does not allow other IP address due to security measures.
VPN/Dyn DNS can be used between these locations so that local IP or DNS host name can be configured to contact other location.

Firewall Configuration Prerequisites while using Static IP for external communication

Port Forwarding
Below ports for HTTP, HTTPS, SIP, RTP range should be forwarded in Router/Firewall to forward relevant traffic to internal UC Server/IP-PBX from external network.

ETERNITY:

Application	Port	Protocol
HTTP	80	TCP
HTTPS	443	TCP
SIP	5060	TCP/UDP both
RTP	8000-9000	UDP

SARVAM UCS:

Application	Port	Protocol
HTTP	80	TCP
HTTPS	443	TCP
SIP	5060	TCP/UDP both
RTP (If only Audio required)	8000-12400	UDP
RTP (If Audio & Video both required)	8000-16800	UDP

Improper port forwarding may result one way speech, failure in registration, no speech etc.

All of the above are default ports and it is highly recommended to change these ports to higher ranges for increasing security of VOIP network.

SIP ALG

SIP ALG stands for Application Layer Gateway, and is common in many commercial routers. It intends to prevent some of the problems caused by router firewalls by inspecting VOIP traffic (packets) and if necessary modifying it. Many routers have SIP ALG turned on by default.

SIP ALG performs the SIP Transformation on Inbound and Outbound SIP Packets in various Headers. SIP ALG changes the Port, Connection Information IP, which leads the Server or Client to respond incorrectly to infinite destination addresses.

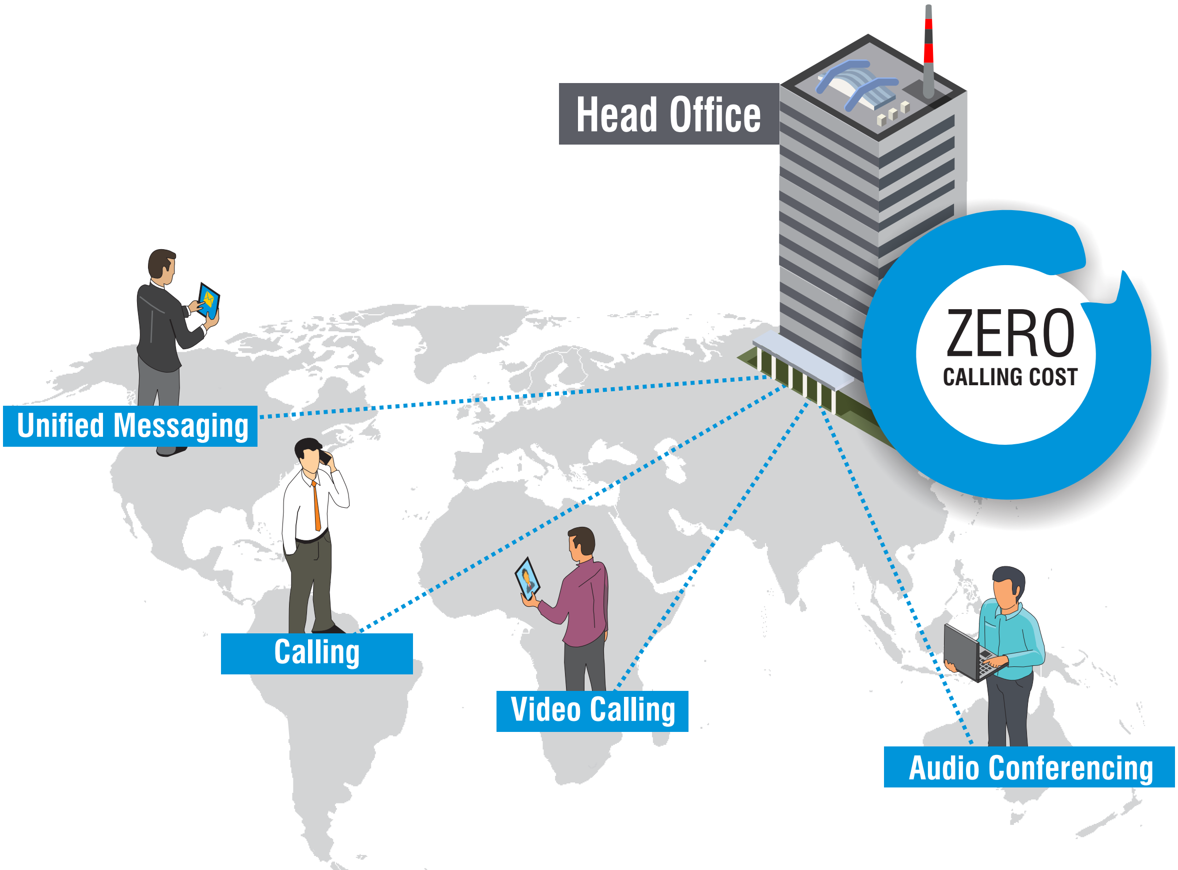
To avoid this, it is very important that your Firewall must not perform SIP Transformation and for Firewall to handle SIP traffic properly, SIP ALG must be disabled.

Other Setting

NAT setting in Router/Firewall must be disabled which may also result in modified SIP packets received from outer network. This can cause one-way speech or no speech/silence issue.

TCP/UDP Flooding must be disabled in Router/Firewall, which can block SIP packets after a certain interval of time.

Note:
This document only gives a brief idea about network setup required while using Voice over IP. MATRIX shall not play any part in configuration or troubleshooting of Router/Firewalls. Additionally, MATRIX shall not provide any brand recommendation of Router/Firewall or WAN interface. Consult Network Administrator while setting up your voice network.



ABOUT MATRIX

Matrix is India based leading manufacturer of IP-PBXs and Gateways for small to large enterprises. Matrix IP-PBX is an integrated communication solution offering universal connectivity with unique design and encompassing advanced features for the businesses of all sizes. Matrix IP-PBX offers benefits of reduced communication costs, seamless connectivity and simplified management for small to large enterprises, institutions, call centres, hotels and many other industries through industry specific solutions. With the global presence in more than 30 countries through an extensive network of more than 500 channel partners, Matrix has gained customer trust and admiration across the world and has won several awards and recognition for its innovative products and processes.

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