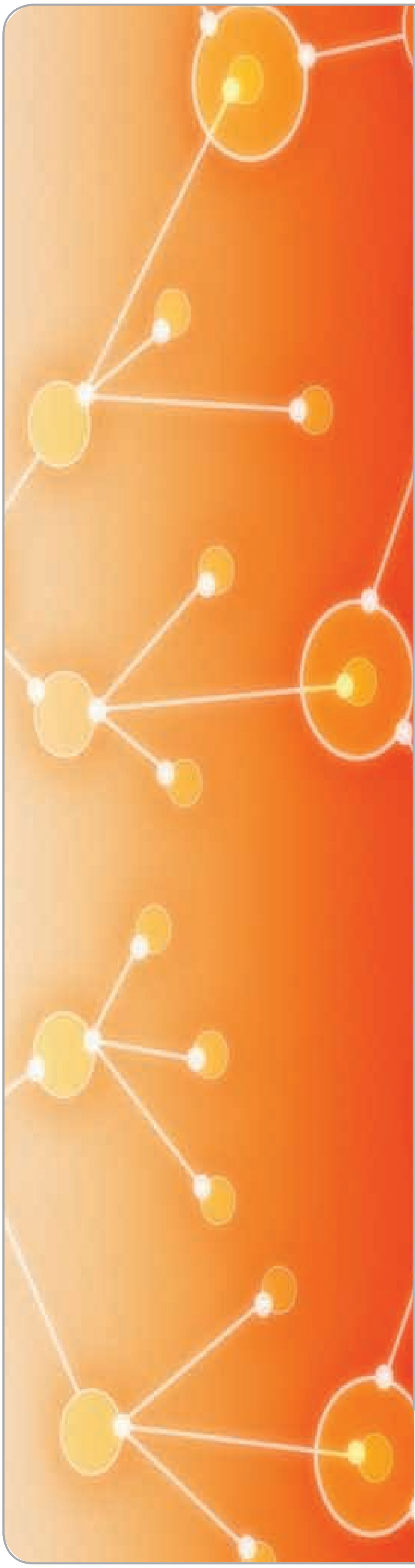


WHITE PAPER

IP Telephony

Ensuring your Structured Cabling
is Suitable

ADC KRONE

Ensuring your Structured Cabling is suitable for **IP Telephony**

Discussions on the impact that a network's structured cabling system has on VoIP operation.

IP Telephony, which includes the commonly known Voice over Internet Protocol (VoIP), is usually introduced into an enterprise as a cost saving measure. This is part of the convergence of data and voice (and video) on the local network so that it is under the control of the enterprise rather than relying on outside specialists. To implement this successfully all components including the network cabling infrastructure, need to be evaluated to ensure the voice quality of the 'telephone' system will not suffer.

How Does VoIP Work?

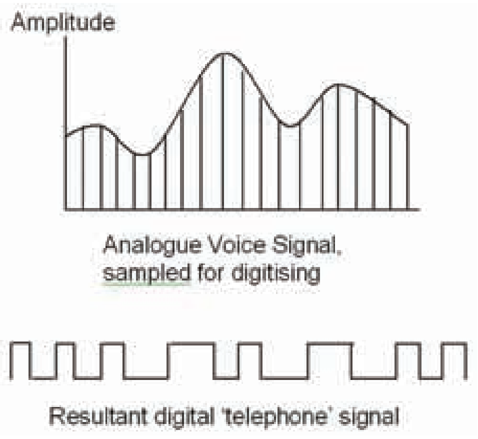
There are three stages in making VoIP work.

First is the conversion of the analogue audio signal into a digital signal by an A/D converter (or codec) at the transmitter end.

Second is the breaking up of the digital signal into packets of data then sending these IP packets to the receiving IP telephone via the network.

Third is the conversion of the digital signal at the receiver using another codec back to analogue audio for the listener.

Speech requires a constant stream of packets, unlike data that can accumulate packets and send them in bursts. To maintain reasonable quality of the conversation, the IP voice packets cannot take too long to arrive at their destination and they must arrive in the correct order.



Transmission Delays

There are four main delays that could affect a VoIP signal:

Propagation Delay is the time taken for the signal to travel from the transmitter to the receiver. If the signal takes too long to arrive, conversation clashes will occur.

Transport Delay is the time taken to pass through each networking device. Every switch, router, traffic shaper, firewall, and hub adds a small delay. For unintelligent devices like hubs the delay is constant, but for intelligent switches the delays increase or decrease as the levels of other traffic on the network increase or decrease.

Packetisation Delay is the time taken to convert the analogue signal into a digital signal and vice versa through the coder/ decoder (codec). Different codecs have different data transfer rates and packetisation delays.

Jitter Buffer Delay is the time taken to queue inside a jitter buffer. Rather than converting VoIP packets directly back to analogue when they arrive, a jitter buffer collects packets arriving at irregular times, ensuring they are in the right order and then sending a smooth stream to the listener. If packets were allowed to be assembled in the wrong order the conversation would become almost unintelligible.

Table 1 Packetisation Delay

Code	Date Rate kbps	Packetisation Delay mS
G.711	64.0	1.0
G.729	8.0	25.0
G.723.1m	6.3	67.5

Quality of Service

Quality of Service (QoS) are software protocols designed to speed VoIP packets through the network system by informing communications equipment that these packets have priority. There will always be some latency (ie. transmission delays) through the network as introduced by switches or different data paths that cause packets to be delayed and arrive out of sequence. So for VoIP, a method of maintaining the constant flow of voice packets in the correct order is essential.

This is partly handled at the receiving end, by the jitter buffer. This buffer cannot be too large, as this itself would introduce an unacceptable delay. Buffer delays are therefore usually only between 20 - 40 milliseconds.

If a packet arrives at the buffer too late to be inserted in the correct order, it is discarded. If a packet is corrupted due to bit errors when it arrives, it is also discarded and there is no time for it to be retransmitted. The sound contained in the discarded packets is not heard and if too many packets are discarded, the conversation becomes disjointed and eventually unintelligible.

Measuring Call Quantity

The measuring of call quantity is usually done subjectively. Simply ask a lot of people to listen to their telephones and rank their perception of the User Satisfaction in say 5 steps from “5 = Very satisfied” to “1 = Totally dissatisfied” (or “0 = Give me back the old system”).

However, there has been considerable progress towards objective measurement systems. For example:

PSQM - Perceptual Speech Quality Measure
(ITU P.861)

MNB - Measuring Normalised Blocks
(ITU P.861)

PESQ - Perceptual Evaluation of Speed Quality
(ITU P.862)

PAMS - Perceptual Analyses Measurement System (British Telecom)

E-Model - A computational model for use in transmission planning
(ITU-T G.107)

Most of these measurements are good in test labs but they are not well suited to assessing call quality in a private data network. The E-Model is the best suited method of measuring call quality and there are software packages available for those that want an objective rather than a subjective measure.

Bit Errors Cause Real Problems

Bit errors will cause IP voice and data packets to be discarded which in turn leads to QoS problems and listening quality problems. Because of the real-time nature of IP Telephony, lost data is never recovered. The luxury of several re-transmissions via TCP applications is not available for VoIP as it is for computer data transfers. Bit errors are introduced into the system through faulty equipment, incorrectly installed structured cabling systems, mismatched cabling components and patchcords, and by external noise sources.

After the system is installed, faulty equipment causing bit errors is easily replaced or repaired and external noise sources can usually be traced and often eliminated. But the cabling infrastructure is not so easily replaced, so it is vitally important that it is installed correctly and tested to ensure there are no situations where the physical cabling is likely to cause Bit Error Rate (BER) problems.

ADC KRONE and all other major component manufacturers say to stay away from external noise sources when installing structured cabling systems. However, noise that is created internally within the cabling system is very much dependant on the quality of the cable, connectors and patch cords as well as the installation pathways and installation practices used by the installer.

VoIP Power

For an IP phone to work it requires a source of power. Currently there are three methods of supplying power; switch supplied power, in-line power, or external power packs.

Switch supplied power comes from the network switch where power is sent down unused pairs and picked off at the VoIP telephone. This mandates that all four pairs of the cable are terminated and available at each end.

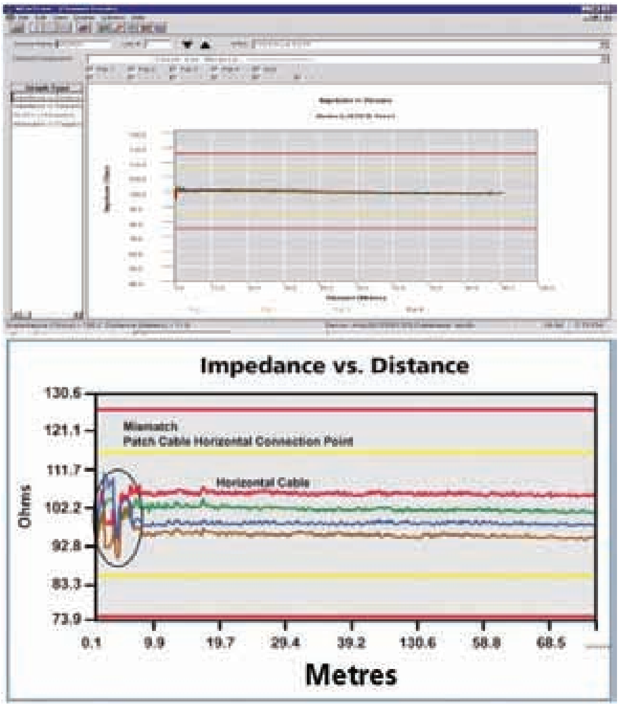
In-line power for an IP telephone is sent down the same pairs that are used for communications and picked off at the receiving VoIP telephone.

Both these methods require the switch equipment to generate and connect the telephone power to the wire pairs. This is usually an option fitted to a switch or a dedicated 'mid-span' device inserted in between the switch and network connections inside the Telecommunications Room.

Note: Be aware that some computer Network Interface Cards (NIC) cannot tolerate voltage on communication pairs.

External power is usually supplied by a power adaptor or "power pack" that is connected to the 240-volt supply at each and every IP telephone. This is usually the least favoured option, but it may be suitable for some smaller sites.

ADC KRONE recommend, for convenience, that IP telephones use either switch supplied power or an in-line supply (or a mid-span) supply method.



Passive Testing

ADC KRONE is committed to providing the best economical cabling system with the lowest BER.

All ADC KRONE Warranted Class D (Category 5e) and Class E (Category 6) installations are tested to the latest international standards using the highest accuracy Level 3/4 field testers. This applies to 100% of the installed runs.

The measured parameters of NEXT, Insertion Loss (attenuation), DC Resistance, Return Loss, Propagation Delay and the calculated parameters of ACR, ELFEXT, skew, as well as all the Powersum calculations PSNEXT, PSACR, PSELFEXT, are all recorded to prove compliance and then presented to the customer for future reference.

Active Testing

What ADC KRONE did initially, as an industry first was to conduct an additional random 10% extra testing on the installed plant focusing on the impedance matching of the components and the installation practices used on site. This gave the customer a second check on how well the job was installed and the ability to confirm the issue of a TrueNet® Warranty.

ADC KRONE is now able to test the actual installed network. We no longer just do 10% at random. We can now test all cabling and connected SNMP-enabled active devices. This testing can be done on request for any Category 6 TrueNet® warranted site.

By migrating to this form of active testing ADC KRONE have also migrated further up the 7-Layer OSI stack. No longer are we measuring just simple active parameters like CRC or FCS errors, jitter, over/under sized packets. We can also see things like capacity and configuration issues, collision domains, incorrect subnets and duplicate IP addresses.

ADC KRONE can audit existing installations to help our customers better understand their current baseline and what productivity is being lost. This enables customers to make better choices of what needs to be done in order for their network to be more efficient and ready for IP Telephony.

PBE for IP Telephony

When ADC KRONE originally invented Patch-by-Exception (PBE) we took advantage of the ADC KRONE patented disconnection contact technology. PBE installations have now evolved to be ideally suited to IP Telephony applications. Because IP Telephony is a data network application, all of the changes to the "Telephone System" will be handled through the software of the IP Telephony Switch in the MIS Equipment Room. There will be no need to physically alter patch cords at a cross-connect vertical in the Floor Distribution for any moves, adds or changes. The usual patch cord mess at the cross-connect will be eliminated forever.

As all network managers and technicians know, the biggest problems in a network usually come from the patch cords in a patching field. By eliminating the patch cords there is a tremendous saving not just in reduced initial capital costs but also in system management and operational fault finding.

PBE uses ADC KRONE disconnection modules such as the Category 6 HighBand® 25-pair or the Ultim8® 10-pair module to hardwire the required jumper field that is then tested for continuity and Class E performance. This gives the customer the assurance that their network will work for both IP voice and data applications.

Now, the real advantage of the PBE system is that if a change is needed that cannot be fulfilled by software switching (eg. between 2 different switches or switch systems) then it can still be "patched by" a physical patch cord as an "exception" to the normal system. Later on when the Patch-by-Exception requirement no longer exists, the system automatically reverts to its original configuration simply by removing the patch cord.

ADC KRONE Recommendations For IP Telephony

Considering that the structured cabling system is considered to be the most time consuming item to install and repair or replace. It should therefore be carefully designed and selected with the appropriate warranty and technical support. An endorsed installer should thoroughly checked during installation for practices that may contribute to non-compliance. Prior to hand-over, the system should also be tested to ensure compliance with the relevant specification in the building contract.

For IP Telephony to be successfully implemented;

1. All four pairs of the cable must be connected in a structured cabling system.
2. The cabling infrastructure should be designed as a Patch-by-Exception installation in the Floor Distributor of new and refurbished installations.
3. IP Telephony power should be switch-supplied as either an in-line or mid-span device utilising equipment and cabling infrastructure which is 802.3af compliant.
4. ADC KRONE TrueNet® Category 6 Patch-by-Exception installations offer optimum capital cost benefits and ongoing operating cost reductions, all installed and tested to give the customer maximum benefits on their IP Telephony system.

NOTE

There is a difference between IP Telephony and VoIP.

IP Telephony usually uses secure IP links like those found inside a single enterprise using a structured cabling system. It can also extend outside the enterprise using dedicated lines linking two enterprise centres. On the other hand, VoIP often uses the unsecured, unmanaged or PSTN (Public Switched Telephone Network) eg. the Internet.

IP Telephony can also deliver the increased functionality and features as seen in modern PBXs. Standard VoIP systems would usually not have these features.

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www.adckrone.com/in

Corporate Office and Factory: P.B. No. 5812. 10C II Phase, Peenya, Bangalore – 560058 INDIA
Tel: +91 80 28396101 Fax: +91 80 28396104
TOLLFREE: 1800 425 8232
OFFICE LOCATIONS: BANGALORE, NEW DELHI, MUMBAI, PUNE, CHENNAI, HYDERABAD

ADC Telecommunications, Inc., P.O. Box 1101, Minneapolis, Minnesota USA 55440-1101