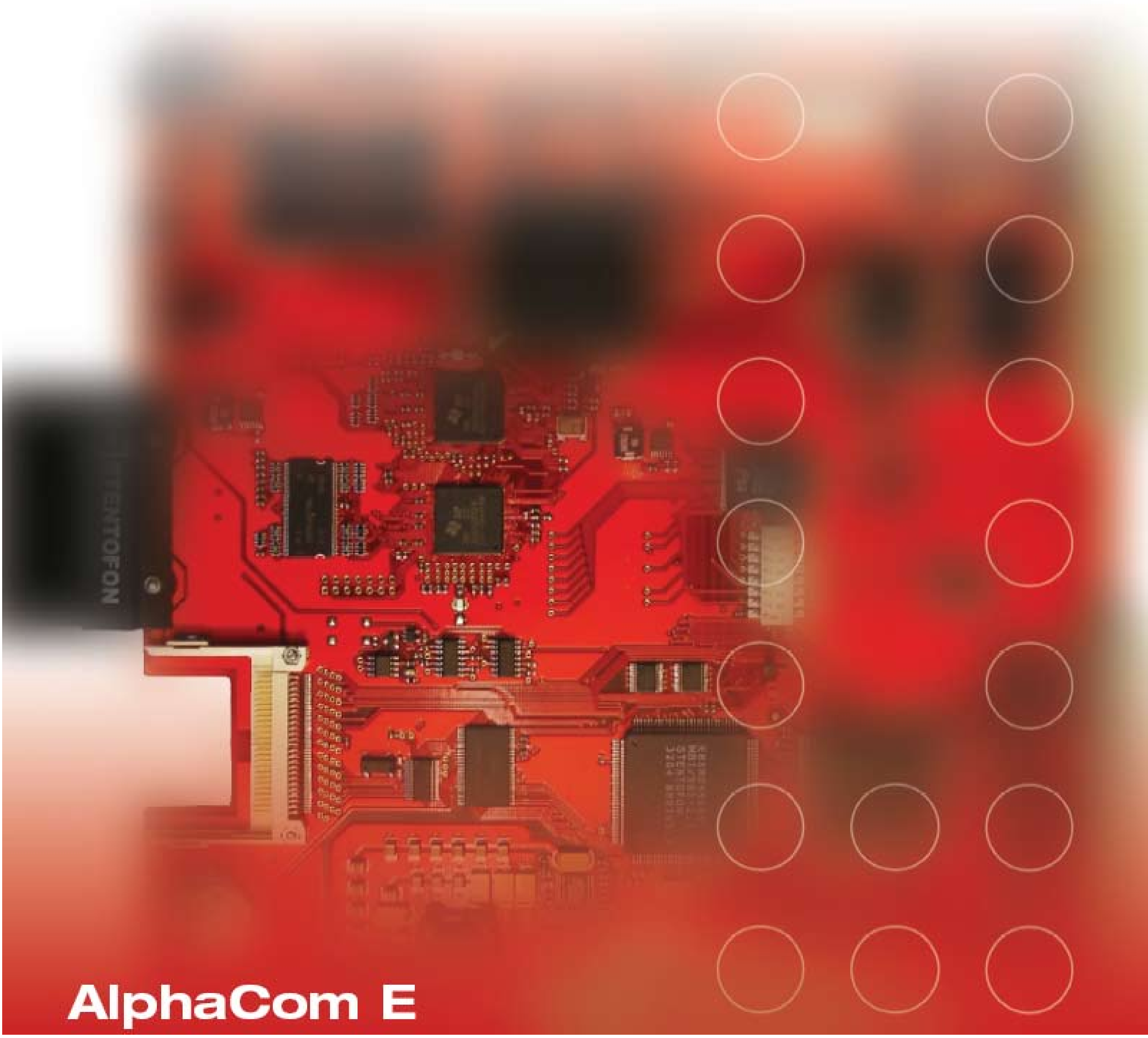

AlphaCom E

FAQ

Frequently Asked Questions and Answers

STENTOFON
ZENITEL GROUP

WhenCommunicationisCritical

STENTOFON AlphaCom E

FREQUENTLY ASKED QUESTIONS

1. Can AlphaCom E provide IP to a station?

AlphaCom E will be able to provide IP to stations. IP stations are part of the next AlphaCom E release.

2. Should I send my critical security system over a VPN? Is it reliable?

VPN provides a very cost efficient way of implementing a WAN service, However, as VPN cannot guarantee a 99.999 % network and bandwidth availability in the Internet, care and consideration should be taken on the criticality on bandwidth availability for the services using the VPN service.

It is possible to build an Intercom over IP service over a VPN WAN link. This will provide a very cost effective solution. As bandwidth availability is not guaranteed and packet loss delay may vary, VPN is not suited for critical communication that must always work. However, if the service can live with a 99% availability VPN, is a good choice.

3. By using IP to connect over intercom network – will we be open to viruses?

The AlphaCom E does not require an anti-virus protection. AlphaCom E does not receive emails nor email attachments, and download of files are restricted to software upgrades. All software upgrade files to AlphaCom have a digital signature to hinder virus infections. The probability of a virus infecting AlphaCom E is thus very small.

Definition virus:
A virus is a program that is designed to infect multiple files on a single computer. It cannot infect other networked computers without human assistance. It will spread to other systems by way of an infected floppy disk, a (infected) shared file on a network drive, or by manually sending the infected file as an e-mail attachment.

4. Is audio quality and intelligibility affected by packet loss? Will I understand the call?

Yes, audio quality and intelligibility is affected by packet loss. You will be able to understand the call with 3% packet loss. However, audio quality and intelligibility will be degraded.

5. Will I still be able to use my existing AlphaCom equipment with AlphaCom E?

AlphaCom E is backward compatible with existing AlphaCom equipment. You will be able to use AlphaCom E with existing AlphaCom exchanges as well as Touchline and AlphaCom station. In addition you will be able to upgrade an AlphaCom 80 and AlphaCom 138 to an AlphaCom E exchange.

6. When utilizing existing or new fiber backbones, can you use off the shelf Ethernet-to-fiber converter and by which recommended mfg?

Yes, you will be able to use off the self Ethernet-to-fiber converters. Zenitel has no recommended manufacture for this type of equipment.

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7. Form for end user or dealer that specifies and agrees to specify QOS?

Zenitel has made a Network Deployment Guide that helps the dealer and end user engineer the IP network.

8. Is it possible to integrate the AlphaCom E with a standard telephone system?

AlphaCom E uses off the shelf Voice Gateways (SIP) to interface to standard telephone systems.

Zenitel recommends Mediatrix Voice Gateways. The Mediatrix Voice Gateways come in two versions, where one provides ISDN interfaces (BRI) and the other has analogue interface (FXO). Each gateway supports up to 4 parallel calls.

Telephone systems which themselves feature a SIP interface can be connected directly via the IP-network.

9. What is the function of 2 separate IP ports, how do they work if one fails, etc.

The AlphaCom E has two separate IP ports, which allows the AlphaCom E to connect to two separate IP networks. The dual IP ports allow our customers to provide higher security and/or redundancy in their systems.

Higher security – separate management interface
Many customers want to perform all administration and management of their networking equipment from a dedicated management IP network to make it more difficult for hackers to get access. AlphaCom E dual IP ports and integrated firewall allows separation of all administration to a dedicated management interface.

Higher redundancy – dual VoIP ports
The second IP port can also be used to provide an alternative AlphaNet route to the other AlphaCom E exchanges.

Note! It is also possible to use traditional analogue and digital links to make AlphaNet alternative routes.

10. What are the distance limitations of Ethernet?

Traditional Ethernet on CAT-5 cable has a link distance limitation of 100 meters between switches and devices.

Ethernet supports a variety of other link types such as wireless and multi mode-, single mode fiber. With a fiber interface Ethernet can travel multiple kilometers.

IP can be transported across many different types of networks. These networks (for instance the Internet) can span the globe, so that there is no real distance limitation.

11. Does AlphaCom E support voice recording?

The AlphaCom E hardware is provisioned to support audio recording. Audio recording is foreseen implemented in a later software release.

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12. Is transcoding for IP stations possible when higher bandwidth networks connects to lower bandwidth networks, in order to enable the best possible frequency response on both stations?

During the call setup, AlphaCom E checks the best possible codec to be used for the call. The best possible codec is then selected for the complete call path. No transcoding is needed.

Note! Current version of AlphaCom supports G.711 (3.4 kHz audio) and G.722 (7 kHz audio). Both of these codec's use 64 kbps.

13. Can any IP camera be connected and set up via an IP Intercom station?

An IP camera can only be connected to an IP Intercom stations that supports IP switching. Zenitel will come with IP Intercom stations variants supporting a dual port IP switch.

Any IP camera can be used.

14. Does IP stations support fiber connection?

The Zenitel IP stations support any type of IP transmission network. However, the first variants of Zenitel IP stations have connectors to CAT cabling. To use a fiber link it is therefore needed to use an Ethernet-CAT-to-fiber converter or switch.

15. How do I connect an AlphaCom E exchange to the Internet via an ADSL modem?

A remote AlphaCom E can be connected to a companies main network through the use of for instance a Cisco PIX router. The PIX router will setup a VPN-tunnel to the main site, through which the remote AlphaCom E becomes part of the main network. An application sheet will describe this in more detail.

16. Will a company who utilizes a Satellite in orbit in space to send data to and from their network experience "latency" issues, how will they be addressed so voice quality is not affected too drastically if not at all ?

Many delay factors in a VoIP system are related to the peculiarities of IP and the coding/decoding of audio. In order to limit the data, a number of voice samples are collected together and sent for instance once every 10 msec. As it is not guaranteed that every package uses the same time to travel from source to destination, a jitter buffer is implemented, another small delay factor. And then there is the time it takes for the package to propagate through the network. A packet sent across the Internet with a source and destination address within Europe takes about 25-40msec, while a packet sent across a transatlantic link takes about 100msec. All these delays added fall within the 300msec round-trip delay; this is the maximum for a voice communication link which is deemed good.

A satellite in a geo-stationary orbit is 40,000 km above the earth. Radio waves travel at 300,000 km/sec. The one way travel time for these radio waves for a single up/down hop is therefore 270 msec. As this is determined by the laws of physics there is nothing that can be done about this. An audio link which involves a satellite link will always suffer this magnitude of delay, whether it is a VoIP link, or a traditional analogue audio link. The upside of this is that the additional delay which is caused by 'IP' is negligible, meaning that a VoIP link is not noticeably worse than an analogue link traveling this same route.

17. Will it be with possible on our new AlphaCom E that one can handle the capability to email or send a page containing event history, alarms, or supervisory trouble shooting data if a customer wanted to know if it could do this? How would this be done?

Yes, the easiest way for the customer to collect logging information is to connect AlphaCom E to a Syslog server like www.kiwisyslog.com . The Syslog server will collect logging information from all AlphaCom E's as well as other nodes in the network. In the Syslog server, you can setup rules for sending emails, reports etc based on special event criteria.

AlphaCom E provides three types of logs. These are:

Technical log	*Includes technical events*
Statistical log	*Includes call detail date like calling parties, VoIP information,*
User log	*Includes custom defined events*

In addition the customer can retrieve and search the AlphaCom logs via AlphaWeb.

18. Will our IP Sub Stations be IP addressable?

Yes

19. Like our hard wired AlphaCom of the past and present, how will Line Supervision and Fault Monitoring be done when one is exclusively utilizing a Fiber Back bone and Cat 5 only? Will the tone test still be done the same way?

AlphaCom E uses keep alive messages to check if remote nodes and stations are operational.

AlphaCom E comes with an extensive Supervision and Fault Monitoring package. For instance each call and for each AlphaNet link, statistics will be generated that records delay in IP network, packets that is lost and packets that experience high jitter delay. The statistics reports can be retrieved via Alphaweb, a Syslog server or a local printer.

Tone tests are still supported the same way by looping a tone between the exchange and the speaker-microphone in the station.

20. Has there ever been any thought of making an IP station with a built in IP CCTV Camera, recently an A & E Consultant asked if I could inquire on this as he mainly focuses on large Sports Stadiums ?

AlphaCom E support for CCTV is under consideration.

21. When does the first AlphaCom IP station come?

See main TGs 2006.

22. What kind of station will this be; wall/desk, vandal proof, weather proof, etc?

The first Zenitel IP stations will be IP substation kits and IP vandal proof substations.

23. What does eavesdropping really mean?

Eavesdropping means that an intruder/hacker is able to listen into a call.

24. Is there any easy way to remember the different bandwidth requirements and audio codec'?

See AlphaCom E Network Design Guide.

25. What exactly is SNMP MIB II (RFC 1905-1907)? A specific standard?

Simple Network Management Protocol (SNMP) is a set of protocols for centralizing the management of IP networks.

It is used for both collecting information from and configuring the full range of network devices, including servers, printers, switches, hubs, routers, firewalls, wireless access points and individual workstations. A wide variety of information can be collected, ranging from a server's CPU (central processing unit) usage level to its chassis temperature.

An SNMP device structures its records in a set of MIBs (Managed Information Base), where the SNMP MIB II includes information such as the nodes CPU status, memory status, active tasks etc.

26. What's the benefit of low latency switching?

AlphaCom E's low latency switching design introduces very low delay in the VoIP path, providing high audio quality. AlphaCom E transit exchange uses less than 1 ms, to forward an VoIP packet.

27. What is PING?

PING is a mechanism to measure network delay between two nodes in an IP network. This is done by sending a PING messages between the two nodes that are looped back. The time it takes to send the message to and from is measured.

28. What is a codec?

A codec is a technology used to make a digital representation of an audio stream.

It is usual to divide the different codec used for VoIP in:

Low bit rate codecs	*- Use less than 15 kbps and supporting 3.4 kHz audio*
Telephone codec (G.711)	*- Uses 64 kbps and supporting 3.4 kHz audio*
Wideband codec (G.722)	*- Uses 64 kbps and supporting 7 kHz audio*
HiFi codec (MP3, 1 bit, CD)	*- Supporting over 15 kHz audio*

AlphaCom E uses 1 bit codec internally and G.711 and G.722 in AlphaNet and SIP.

29. What is a jitter buffer?

The objective of jitter buffer design is to keep the buffering delay as short as possible, while minimizing the number of packets that arrive too late to be used.

VoIP transmission delay varies quickly and with significant amounts over time due to queuing effects in the IP network, causing a delay jitter. The jitter present in packet networks complicates the decoding process because the decoder needs to have packets of data readily available at regular intervals to produce smooth, continuous speech.

30. What is an adaptive jitter buffer

The objective of jitter buffer design is to keep the buffering delay as short as possible, while minimizing the number of packets that arrive too late to be used.

An adaptive jitter buffer is able to automatically adapt to the current network behavior to optimize on reducing delay without loosing too many packets.

AlphaCom E has adaptive jitter buffers.

31. Backwards compatibility. Is this a feature you plan to maintain long-term? (A statement on company policy/objective to maintain this here would be good)

Yes, Zenitel is committed to provide backward compatibility for the AlphaCom and AlphaCom E systems.

32. What is an RFC?

An RFC (Request For Comments) are an Internet standard. All RFCs are posted on www.rfc.org by the Internet Engineering Task Force (IETF).

33. What additional Syslog functionality is bolted-on/purchased?

It is possible to connect AlphaCom to a Syslog server like www.kiwisyslog.com . A Syslog server can be used to collect log information from multiple nodes of same and different type.

In addition to the log and event collections most Syslog servers comes with many advanced logging functions like network management and supervision, email notifications, SNMP and more.

Note! AlphaCom E has an extensive log package in the basic software. See AlphaCom E System Management and Operation manual for more information.

34. Can you explain the computing term 'stateful firewall' or a stateful inspection?

The difference between a stateful firewall and a basic firewall is in how they inspect the packets before allowing is into the network.

A basic firewall looks at one packet at a time, and considers it in isolation in order to make a forwarding decision. This is called packet filtering

A stateful firewall uses stateful inspection taking the basic principles of packet filtering and adds the concept of history, so that the Firewall considers the packets in the context of previous packets.

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35. If a hacker somehow gains access to our network can he use this access to get into the company/organizations network?

The AlphaCom E does not support direct packet forwarding between its 2 IP-ports. IP-traffic which enters the exchange is evaluated and processed by the exchange. The exchange either views it as intercom related data (commands), or as audio; if the exchange does not understand the IP-packets as AlphaCom related it will ignore the packet. It is therefore not possible to gain access to a network via the AlphaCom exchange.

36. Will we be able to identify the 1st dropped call?

In an AlphaNet a primary and an alternative route can be defined for each possible combination of end-exchanges in a conversation. As this primary/alternative routing can be defined for each transit exchange along the route of a call, the number of possible routes in a large network can be enormous. A call will always by preference be set up via the primary route. If somewhere along the route part of the primary route is not available, alternative routing will be used. A call will therefore always be set-up, unless there are not enough resources available, for instance because all available audio links between exchanges are assigned to other calls.
If a call has been set-up, and a cable between exchanges gets broken, that call will be cancelled. The system will not automatically try to setup a new call. By using a combination of the error log and the statistical log it will be possible to identify which call was affected.

37. Are there any plans to do a wireless IP substation?

Also see the answer related to IP-stations in general. A STENTOFON wireless IP-station is not planned at the moment. But if the requirement is to bridge a distance where there are no copper or fibre cables available, it is possible to bridge that distance using commercially available products to implement a WiFi-network. When the first IP-intercom station is available Zenitel will also release an application sheet showing a possible configuration.

38. Is the licensing based on an annual fee or one time fee?

Currently the licensing is based on a one time fee. Zenitel will communicate any changes in its licensing scheme well before such changes would take effect. Already sold 'one time fee' licenses will never be affected by such changes.

39. What is SIP?

SIP is one of the standard protocols for IP-telephony. The other main protocol in use is H.323. H.323 is the older protocol and originated in the telecommunications industry, SIP is a computer industry protocol. The expectation is that SIP will be the protocol mainly used in the future as it is easier to implement and debug. Many national telephone networks will be built upon the SIP protocol.

40. This system can be connected to the telephone system, so what is the Telco-companies part of the configuration?

It has always been possible to connect an AlphaCom exchange to a telephone network through the use of the PNCI. The implementation of SIP gives us even better connection and integration possibilities to PABX's and national carriers.
Direct connection to a national carrier network is of importance when the AlphaCom is used as PABX, or when it is necessary for an operator to be able to call, or forward calls to for instance police or fire brigade. Connection to a PABX is of importance if the PABX system is used for the general communication needs of a company, while the special requirements are implemented using AlphaCom and its intercom stations, for instance industrial stations or stations in an operating theatre.

41. Can this system be used like "Skype" and similar systems?

Skype is free software which can be used for people to communicate via computer networks. It does not support any kind of QoS. Companies which have an AlphaCom installed can also use it as a toll-bypass communications system just as Skype, but in addition it can also support QoS.

42. When the delays and its variation are too big, how can we measure that before the installation?

By sending a PING message (see above) to a IP-address it is possible to measure the time it takes for a packet to travel to and from the destination. On a Windows PC it is possible to enter a PING-command from the 'Command prompt window'. Four messages are sent, and the result is shown with a minimum time, average time and maximum time. The difference between minimum and maximum gives indication of the variation. Note that it is normally best to do this test a number of times. If out of a number of tests one of the maximum times is wildly out it can be ignored. If the variations are constantly very high, or the result shows a large number of dropped packets, the link should not be used to implement AlphaNet over IP.

43. Does E- series system support all the functions in an AlphaCom over the network, if not, then which one are not supported?

AlphaNet over IP supports all functions which are supported by the 2 other AlphaNet configurations, using the AE1 or AGA cards. In addition AlphaNet over IP supports dynamic AphaNet, a more efficient way of using defined audio links.

44. What is Dynamic AlphaNet?

Dynamic AlphaNet uses the switching capabilities of an IP network. An AlphaNet connection on an IP network is just an audio stream from one IP-address to another IP-address. It is not a physical link. Therefore, when the conversation is finished the AlphaNet port is free. A next conversation on that port can be connected to an entirely different IP-address, e.g. a totally different AlphaCom exchange. So even in a very big network, even if every exchange only had a single AlphaNet port defined, it would still be possible to make a call from any exchange to any other exchange, while in a 20 exchange network it would still be possible to have 10 AlphaNet calls simultaneously. The in a static AlphaNet most cost-effective system (a long daisy chain of exchanges) would in 18 exchanges require 2 audio ports and in the other 2 exchanges 1 port. A call from exchange 1 to exchange 20 would tie-up all AlphaNet resources and in addition would in practice take far too long to set up.

45. What kind of applications have already been made with the E- series?

The AlphaCom E has been used in all segments in which the AlphaCom has been used as well. As the AlphaCom E is built on the AlphaCom application software, all previous features which were implemented to serve the different segments are still available. On top of this there is now IP-connectivity, remote programming and maintenance and the vastly increased processing power of the new AMC-IP card.

46. When are licenses needed and when not?

There are 2 types of licenses at the moment, AlphaNet and multi-module over IP and SIP trunking. Implementation of AlphaNet or a multi-module configuration with AE1 or AGA cards does not require the purchase of a license. Also the use of a PNCI to implement a trunk line towards a telephone system does not need a license. In the future more licensed functions will be implemented.

47. Trunk lines, what does it mean, how to measure the telephone traffic and need?

A trunk line is a line between communication exchanges. It is a shared resource, which is assigned to parties needing to communicate, for the duration of the call. When the call is ended, the trunk line is released for other parties needing to communicate.
The unit of traffic is the Erlang. In practise it is used to describe the total traffic during an hour. A single channel, used continuously represents 1 Erlang of traffic. The number of trunk lines required is a statistical calculation. It depends on the average length of a call, the number of people who are allowed access to the trunk, and the willingness of a company to encounter a busy line. There are calculators available on the Internet (for instance on www.erlang.com/whatis.html), where it is possible to enter a number of variables and which will assist in calculating the number of trunk lines needed.

48. What in a practice makes this system better than the other VoIP- systems on the market?

The AlphaCom E is better than other VoIP-systems on the market, because it is builtfrom the bottom up to support CCoIP. The complete design is focussed on being able to deliver communication when they are needed.
Another reason is that the fact that the AlphaCom E is backward compatible with the AlphaCom has not stopped Zenitel from designing a processor card which supports native IP. In many cases, the VoIP support is something that is added, while not doing anything about the architecture of the system. The AlphaCom E can be viewed as a dual architecture exchange (2 exchanges into 1). Forward and backward compatibility each served by their own architecture.

60 years of quality communication

From life-saving operations at sea to security and communication in a world perspective.

In 1946 a Norwegian by the name of Otto S. Knudsen began producing life saving marine radios in Trondheim, Norway, leading to the production of radio sets, car radios and eventually intercom systems. This was the start of a company gaining international significance for its groundbreaking innovation, the STENTOFON intercom system.

Production of intercom systems began as early as 1946 but really picked up speed in 1960 with the launch of the STENTOFON ST30 intercom system. Export contacts were being established with a number of countries, and offices were opening abroad including Denmark, Sweden, Germany, France, USA, UK, China and Australia.

The first Pamex models were released in the 1970's and intercom development went into overdrive in the early 80's with the microprocessor controlled systems. The primary product being manufactured during this time was the Pamex MPC/TouchLine that had much success until it was superseded by the AlphaCom system. The STENTOFON system is used in wide ranging applications around the world.

The 90's witnessed the alliance of the three strongest and longest-established Norwegian brands in the field of internal communication. STENTOFON and Ring-Master merged in 1997 to become Stento ASA, and the M100 system produced and marketed by Philips was acquired in 1999. Stento ASA merged with SAIT Radio Holland in May 2000 and the company changed its name in 2001 to become Zenitel.

All through these years, the STENTOFON brand hasn't stopped seducing more prestigious and demanding customers such as Capitol Hill in the United States...

Zenitel has world-wide representation

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A100K10320 - August 2006