



IHM VoIP products



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1 IHM VoIP Product Range

IHM offers a range of various Voice over IP products. The product range includes Headset boxes, Dispatcher terminals, Radio Base Station controllers, and Interface modules for the IHM series of Communications switches. All products are connected to a network via a standard RJ-45 LAN connector with 10/100MB/s interface.

The IP-addresses for the modules are coded via an RS232 connection and a special windows program. All other parameters can hereafter be coded from a PC connected to the same network using a special windows PC program.

From the same windows program, you can also update the module software (Firmware).

Most IHM products are based on IHM's own proprietary protocols, UDP. However, a SIP telephone module is available using standard SIP and RTP telephone protocols.

The modules can support 'Quality of Service'.

As all modules are built using a processor, where the software is made especially for this purpose, it is not possible to pass any data through the modules to any other network connected equipment and likewise it is not possible to be infected by virus. Besides the proprietary IHM protocols the modules for service reasons however answers a standard ping command, further IP-standard commands/functions are not supported.

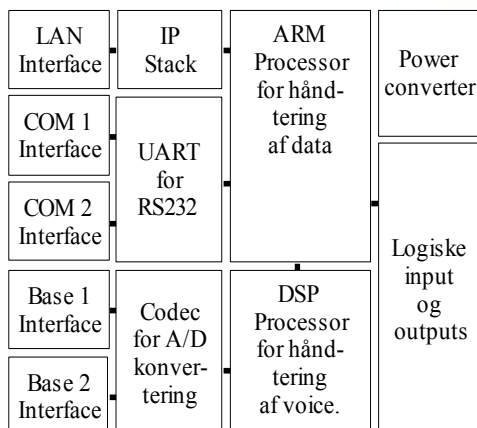
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1.1 Com53 Base Station Controller

Base Station controller holding:

- Ethernet connection
- Audio
- PTT / SQ / RSSI (A/D converter)
- RS232
- 6 Additional Outputs
- 8 Additional Inputs



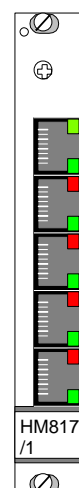
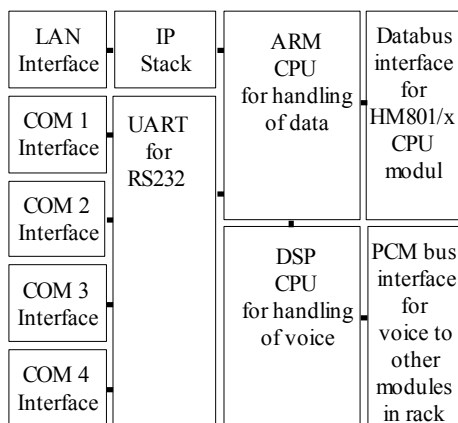
The IHM COM53 is a stand-alone unit, which also can be used for a point-to-point connection with up to two simultaneous voice channels and up to two RS232 data channels typically used as base station remote controller.

1.2 HM817/x

HM817/x is a VoIP module for IHM modular communications switches COM26xx and COM45xx.

HM817/x is available in two versions:

- HM817/1 for communication towards up to 4 x COM53 or 4 x TE1040/3 operator audio boxes or 8 x SIP telephone lines.
- HM817/3 for establishing up to 8 conversation channels between to sites with remote COM45xx systems (remote shelf).





1.3 TE10-39/3 Dispatcher terminal

The IHM TE10-39/3 is a digital desktop radio dispatcher terminal. The terminal in its basic version is configured for VoIP connection to one or two radio base stations fitted with the IHM-COM53 base station controller or VoIP connection to an IHM COM2600 or COM4500 Communications switch.

Desktop dispatch terminal includes:

- 16 programmable keys
- Loudspeaker
- Gooseneck microphone
- Headset plugs
- Foot pedal plug
- Sound card connection
- Two sockets for optional AUX (4W-E&M) connections



1.4 TE10-40/3 Headset Box

The IHM TE10-40/3 is an IP enabled digital audio box for Computer Aided Dispatch systems connected to the IHM-COM4500 Communications switch. The TE10-40/3 hold all necessary interfaces, loudspeaker amplifiers, side-tone filters, limiters and AGC to fulfill Operator requirements in mission critical command and control centers.

The Headset audio box includes:

- Ethernet connection
- Plug for PC stereo loudspeakers
- Headset plugs
- Microphone and Foot operated PTT plug
- Sound card connection
- Three sockets for optional AUX (4W-E&M) connections



1.5 TE10-41/1 Dispatcher terminal

The IHM TE10-41/1 is an IP enabled digital audio desktop terminal for Computer Aided Dispatch systems connected to the IHM-COM4500 Communications switch. The TE10-41/1 hold all necessary interfaces, loudspeaker amplifiers, side-tone filters, limiters and AGC to fulfill Operator requirements in mission critical command and control centers.

The Desktop dispatch terminal includes:

- Built-in Loudspeakers
- Gooseneck microphone
- Headset plugs
- Gooseneck microphone plug also includes PTT input, which can be shared with a foot operated PTT pedal.
- Sound card connection
- Separate Traffic/Monitor mode volume sliders
- Large PTT button
- Three sockets for optional AUX (4W-E&M) connections



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1.6 TE10-41/3 Dispatcher terminal and system controller

The IHM TE10-41/3 is an IP enabled digital audio desktop terminal for Computer Aided Dispatch systems connected to the IHM-COM4500 Communications switch. The TE10-41/1 hold all necessary interfaces, loudspeaker amplifiers, side-tone filters, limiters and AGC to fulfill Operator requirements in mission critical command and control centers.

Furthermore, the terminal includes a communications switch which can control up to 10 IP streams, which can be used towards TE10-39/3 audio terminals, TE10-41/1 audio terminals, COM53 radio base station controllers and RTP streams to voice recorders. The TE10-41/3 can be equipped with a 2-wire analogue PBX/PSTN module.



The Desktop dispatch terminal includes:

- Built-in Loudspeakers
- Gooseneck microphone
- Headset plugs
- Gooseneck microphone plug also includes PTT input, which can be shared with a foot operated PTT pedal.
- Sound card connection
- Separate Traffic/Monitor mode volume sliders
- Large PTT button
- Three sockets for optional AUX (4W-E&M) connections

1.7 COM40

The COM is a communications switch which can control up to 10 IP streams, which can be used towards TE10-39/3 audio terminals, TE10-41/1 audio terminals, COM53 radio base station controllers and RTP streams to voice recorders.

The COM 40 can be equipped with an analog 2-wire analogue PBX/PSTN module.



The COM40 can be delivered either as a stand-alone unit or delivered in a 19" 2U sub-rack including a Mini windows PC and a network switch.

1.8 PC Audio

IHM PC-Audio is a Windows driver making it possible to use USB enabled headsets, loudspeakers microphone and PTT switches connected directly to the Operator PC. The driver also includes RTP streaming for IP recorders as for example ASC neo.

Please note, Windows limits audio devices to one audio device active at the time, you can therefore not use PC in Tudor/Student operator positions.

1.9 RTP streaming to IP voice logger

IHM has made two protocols available for the TE10-xx, HM817 and PC-Audio which can stream audio received and transmitted to an external IP voice recorder.

The two available protocols are IP address based RTP for NICE recorders and IP port based RTP for ASC Telecom recorders.

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1.10 IHM SIP server

IHM has released a SIP server software which works perfectly with the IHM COM4500 series. Amongst others the SIP server includes routing which makes it possible to receive Calls from SCAIP Home Care terminals, which does not include a telephone number.

1.11 IHM SIP Trunk

IHM has made an option for the HM817 module making it possible to connect to SIP Servers and SIP providers. The SIP Trunk module supports amongst others CISCO, BREKEKE, IHM SIP servers and various SIP providers. Please note, as SIP providers may use different protocols and offer different functions, modifications to the HM817 firmware shall be accepted, if the provider or SIP server is not already supported.

1.12 FWT - Fixed Wireless TETRA access terminal

IHM FWT is an access terminal for TETRA networks, providing all the facilities which are made available over the air from the TETRA backbone system. The IHM FWT is available with 1 or 2 Voice channels and it is available for wall mount or 19" rack mount.

IHM-FWT Supports:

- Private Call
- Group Call
- Alarm Call
- Text Message
- GPS Position





2 Voice Quality

The voice quality depends upon several parameters in the IP network and the Codecs used. The network parameters which will have impact on the quality are shortly described in this section.

2.1 Codecs

To ensure the best voice quality IHM products only uses the Voice Codecs G711 μ -law and G711 A-law. These Codecs uses 8000 samples per second. The signal is sampled using 13 bit, and the compression algorithm will change the 13 bits to 8 bits. The data stream required will then be 64Kbit/s.

In addition to the audio data, a header is needed for each UDP IP-Packet. Using UDP packets each holding 20ms samples, will then require an IP data connection of approximately 90Kbit/s in each direction.

A small overhead is required to ensure the optimal voice quality and IHM therefore recommend a bandwidth of 128Kbit/s for each voice channel.

| Codec | Bit when sampled | Bit when compressed | Bit Sample rate |
|-----------------|------------------|---------------------|-----------------|
| G711 μ -law | 13 | 8 | 8000/s |
| G711 A-law | 13 | 8 | 8000/s |

2.2 Bandwidth Requirements

Even that 128Kbits is the recommended bandwidth for each voice channel it might be possible to use a lower bandwidth.

The table below shows the relationship between bandwidth and the number of voice channels in use.

Required Bandwidth Kbit/s

| Number of connections | | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
|-----------------------|-----------------|-----|-----|-----|-----|-----|-----|-----|------|
| | | | | | | | | | |
| Recommended | G711 μ -law | 128 | 256 | 384 | 512 | 640 | 768 | 896 | 1024 |
| | G711 A-law | 128 | 256 | 384 | 512 | 640 | 768 | 896 | 1024 |
| Absolut minimum | G711 μ -law | 110 | 205 | 300 | 395 | 490 | 585 | 680 | 775 |
| | G711 A-law | 110 | 205 | 300 | 395 | 490 | 585 | 680 | 775 |



2.3 Network Jitter

The network will typically have some kind of jitter. This means that the network packets are periodically delayed more or less than the average delay. This could give short breaks in the audio stream, as the receiver has no audio packets to send out in the audio stream. To avoid this problem, IHM VoIP units hold a network jitter buffer. This buffer holds enough packets to continue the audio stream even that no packets are received from the network for a short moment.

The disadvantage of this buffer is that the delay will be increased with the buffer size.

The buffer size is configurable in steps of 20ms up to 160ms. In most cases a buffer of 20ms will suffice to avoid interruptions caused by jitter, and this will introduce the lowest possible delay.

| Configuration | Buffer size |
|---------------|-------------|
| 1 | 20ms |
| 2 | 40ms |
| 3 | 60ms |
| 4 | 80ms |
| 5 | 100ms |
| 6 | 120ms |
| 7 | 140ms |
| 8 | 160ms |

2.4 Multiple audio path

For some applications, there will be more than one audio path open between a radio and an operator position. This could be if an operator position is listening to several base station sites, which are using the same radio channel.

If the delay is different between the two radio base stations sites and the operator position, an echo will be heard by the operator.

To minimize this problem IHM equipment is able to put an additional delay on the audio path with the shortest delay. It is possible to adjust the additional delay to make the two signals arrive with a difference of maximum 25msec.

This requires that no additional varying delay is introduced by the network.

2.5 Latency

The complete latency when using two IHM VoIP units can be found by adding each step in the VoIP communication.

The table below indicates the total latency when the network adds 10ms average delay, and audio packets are sampled in 20ms packages.

| Action | Latency |
|-----------------------------------|---------|
| Converting from analog to digital | 1ms |
| Buffering up a package | 20ms |
| Internal data handling | 3ms |
| Network delay | 10ms |
| Internal data handling | 3ms |
| Jitter Buffer | 20ms |
| Converting from digital to analog | 1ms |
| | |
| Total Latency | 58ms |

If the network jitter buffer is increased to 40ms, the total Latency for one direction will be 78ms.



2.6 Quality of Service

If the network connection used by IHM VoIP product is shared with other network applications, it may be required to use Quality of Service (QoS). With this feature, it is possible to add a priority to network connections using VoIP.

The configuration is setup in the network routers. This ensures that VoIP packets are transferred first, even that a heavy file transfer or video streaming has been initiated.

In case QoS is not configured the file, transfer may course a break in the audio stream.

Routers normally have the option to add priority by specifying the source of VoIP traffic. The priority can be associated with:

- The source MAC address
- The source IP address
- An IP address range
- An IP Port range
- The Ethernet plug in the router

Some routers also have the option to use the ToS (Type of Service) field in the IP package. This field is configurable in IHM VoIP products. The three most significant bits of this field forms the PRECEDENCE, and only this part is Configurable. Six options are available.

| Precedence | Priority |
|------------|-------------------------|
| 0 | Best Effort |
| 1 | Priority |
| 2 | Immediate |
| 3 | Flash |
| 4 | Flash Override |
| 5 | Critical, Used for VoIP |
| NA | |
| NA | |

2.7 Network requirements

IHM VoIP solutions in general requires no more than 150 msec of packet delay on UDP data stream and a packet loss rate less than 0,1%.

Command interfaces on IHM VoIP enabled devices uses retransmission methods to reduce interference from packet loss.

| | HM817/n | TE10-nn/n | COM53/FWT |
|-----------|---|-----------|-----------|
| LAN speed | 10/100Mbit | | 10Mbit |
| Bandwidth | 128kbit up/down per active voice channel | | |
| Latency: | <150msec one direction, 300msec both directions. | | |
| Jitter | < 160msec | | |
| Broadcast | Should not take place in networks with IHM IP modules | | |



3 IP connection principle

The IHM equipment operates in a master/slave configuration.

Establishing an IP-connection between two modules is always done by the master sending a UDP request to a specific IP-address and specific code-able IP-port to locate the pre-coded slave module. When the slave module responds (answers) to the address requesting the connection, a UDP stream between the two modules is establish.

3.1 Keep Alive

Every 2nd second a UDP data package is transmitted from master to slave, who responds simply to verify the connection between the two modules. Included in the same package is also a status update of input and output ports. If the acknowledge is not received, the master retries in up to 8 seconds before the connection is considered faulty.

3.2 Controlling inputs and outputs

Upon a status change on an input port a UDP package is transmitted to the opponent module, who acknowledges the packages. If the acknowledge is not received, the module retries in up to 8 seconds.

3.3 Voice data

When voice is transmitted, this happens via the DSP processor on the module. 20msec of voice is collected, transferred to the ARM processor, whom converts the speech into a UDP package and sends it to the network. At the opponent module, the package is received and converted to 20msec of speech, send to the appropriated audio port. If a UDP packages is missed in any way, it will not be retransmitted.

3.4 SIP protocols

IHM supports standard telephone functions towards a waste number of iPBX systems, for example Cisco, Brekeke, Asterix, Astra MK1 and obviously IHM own SIP server. However please note that IHM does not necessary support vendor specific functions.

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4 Building a network structure

This chapter is written primarily for radio and telecom engineers and not for network engineers, as these may find the following information obvious.

A lot of factors may affect the choice of network structure, e.g. security for third party intruders, safety of data transport, simplicity and the possibility of admittance to services.

4.1 IP-addressing

Each unit in a network holds a unique IP-address, but for every IP-address a subset of addresses is found also called TCP or UDP Ports. Every IP-address can have up to 65536 ports defined. Some of these ports are dedicated for special purposes e.g. port 80 is always for http, which is used for web pages.

4.1.1 IP-address

An IP-address is divided into octets, given an IPv4 address consists of 32 bits, 8 for each corresponding decimal number.

Internal networks addressing is a written standard by Internet Engineering Task Force (IETF), the standard is named RFC1918, and dictates the following IP-addresses is not to be routed on the internet and therefore reserved for private networks:

10.0.0.0 /8 (10.0.0.0 – 10.255.255.255)

172.16.0.0 /12 (172.16.0.0 – 172.31.255.255)

192.168.0.0 /16 (192.168.0.0 – 192.168.255.255)

4.1.2 Net mask

The subnetmask or just netmask decides the size of the subnet, this is done by a bitwise AND-operation of the IP-address and the subnet mask:

| | | |
|------------|--------------------|-------------------------------------|
| Address: | 192.168.1.10 | 11000000.10101000.00000001.00001010 |
| Netmask: | 255.255.255.0 = 24 | 11111111.11111111.11111111.00000000 |
| => | | |
| Network: | 192.168.1.0/24 | 11000000.10101000.00000001.00000000 |
| Broadcast: | 192.168.1.255 | 11000000.10101000.00000001.11111111 |
| First IP: | 192.168.1.1 | 11000000.10101000.00000001.00000001 |
| Last IP: | 192.168.1.254 | 11000000.10101000.00000001.11111110 |

If an endpoint is attempting to contact another endpoint which is not on the same subnet, it will contact its default gateway.

4.1.3 Default gateway

The Default Gateway is the unit in the network that provides connection to other networks. Meaning that if we try to call an address not found within the net-mask, the call is routed via the default gateway. This again means that all signaling to other networks is provided via the Default Gateway. Further you may find different kinds of blockings and IP-address translations. As an example, if a unit from outside calls a specific IP-port, this could be translated to an internal specific IP-address and perhaps another IP-Port. This function is also known as NAT.



4.1.4 MAC address

Besides the IP-address which may be variable, all network enabled products also holds a unique hardware coded address. These addresses are provided by IEEE (Institute of Electrical and Electronics Engineers organization), in such way that each manufacturer is provided a series of addresses. The MAC-address is the identity that Ethernet switches uses to control the IP traffic.

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5 Network types

In the following chapter, we will take a look at different scenarios for configuration of networks. In all examples, we use a system configuration consisting of one HM817/1 and 4 x COM53, as this is the most complex system configuration. Other system configurations will be more or less similar to the example given.

5.1 Providing IP-addresses

There are 3 ways of providing IP-addresses.

5.1.1 Dynamic (DHCP)

If the DHCP server is on the same subnet as the endpoint which is requesting the DHCP assigned IP, it is the MAC of the endpoint. But the DHCP could be located on another subnet, and would therefore be the MAC address of the nearest router/gateway to the DHCP server. The address could change every time the end-point is connected, it depends on the DHCP servers lease time. The end-point will attempt to renew its lease, when it reaches its half-life.

Beside the gateway and netmask, the DHCP server could also offer DNS-, NTP-servers, DNS-search suffix, TFTP server etc.

5.1.2 Fixed IP from DHCP

Using fixed DHCP the unit will upon activation request an IP-address. This request is handled by a DHCP server, which will respond to the request using the MAC-address provided by the requesting unit.

Stored in the DHCP server a list of fixed IP-addresses to corresponding MAC-addresses is found, the DHCP server will therefore always provide the same IP-Address to the same unit.

In the same package also Net-mask and Default Gateway addresses are provided.

5.1.3 Static IP

When using Static IP, the units hold a programmed IP-Address, besides Net-Mask and Default Gateway addresses. Using this method there is no requirement for having a DHCP server.

IHM modules support Static IP only, whereas other service can be provided as fixed or dynamic IP addresses..

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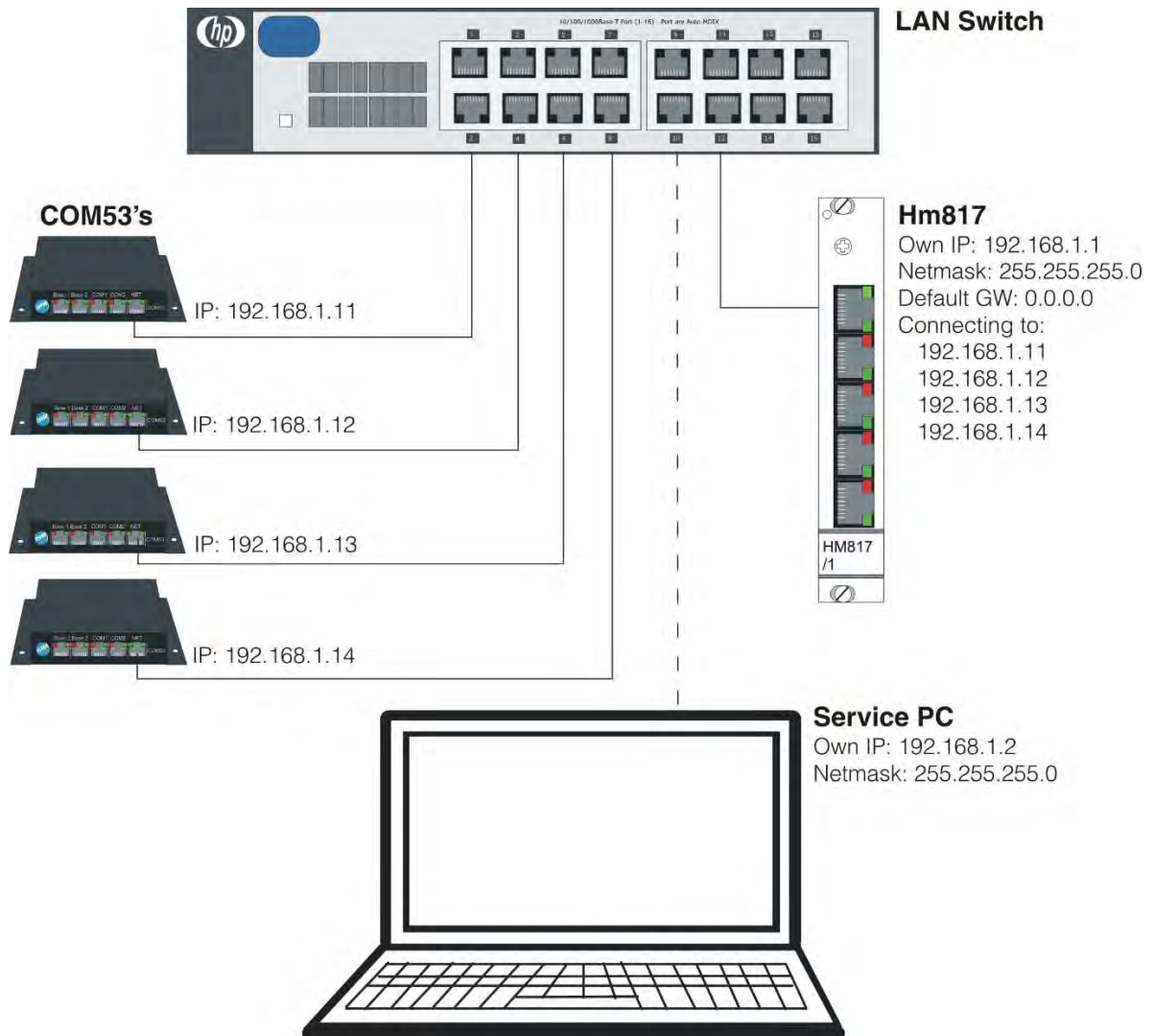
5.2 LAN

In the examples, hereunder we concentrate on internal networks. Internal networks may be spread over a larger geographical area using links or fiber optics. The important issue is the structure of the IP-addressing.

Internal network may also be created using Fiber Optics, MPLS or VPN.

5.2.1 Simple LAN without other users

Also, referred to as a single layer 2, domain.



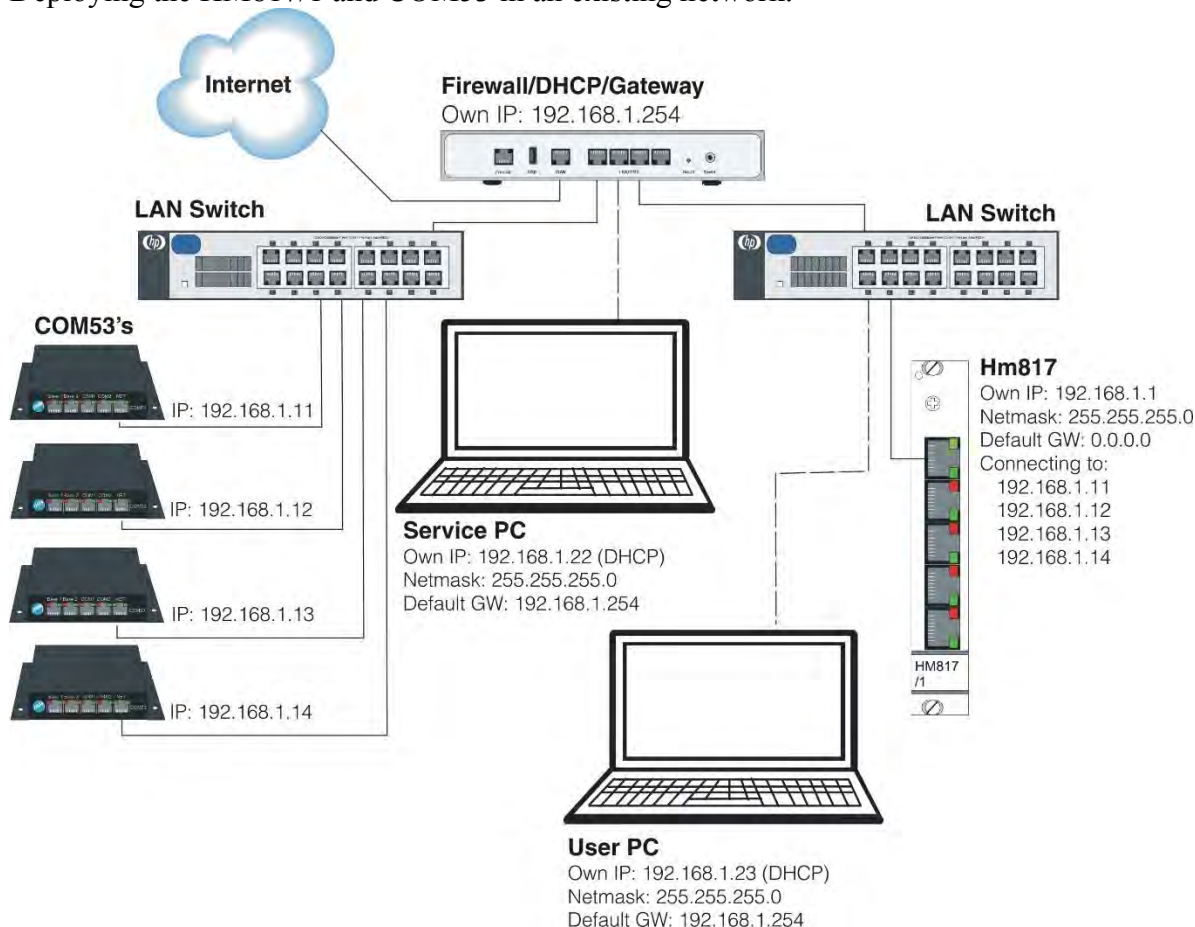
In this configuration, the network is solely used for communications between HM817/1 and COM53. All units are coded with a Static IP-address, no further equipment is required. In this configuration, there is no access from other networks. Also, the PC for configuration needs to be address coded when connected to the network.



5.2.2

Internal LAN without divided IP-space

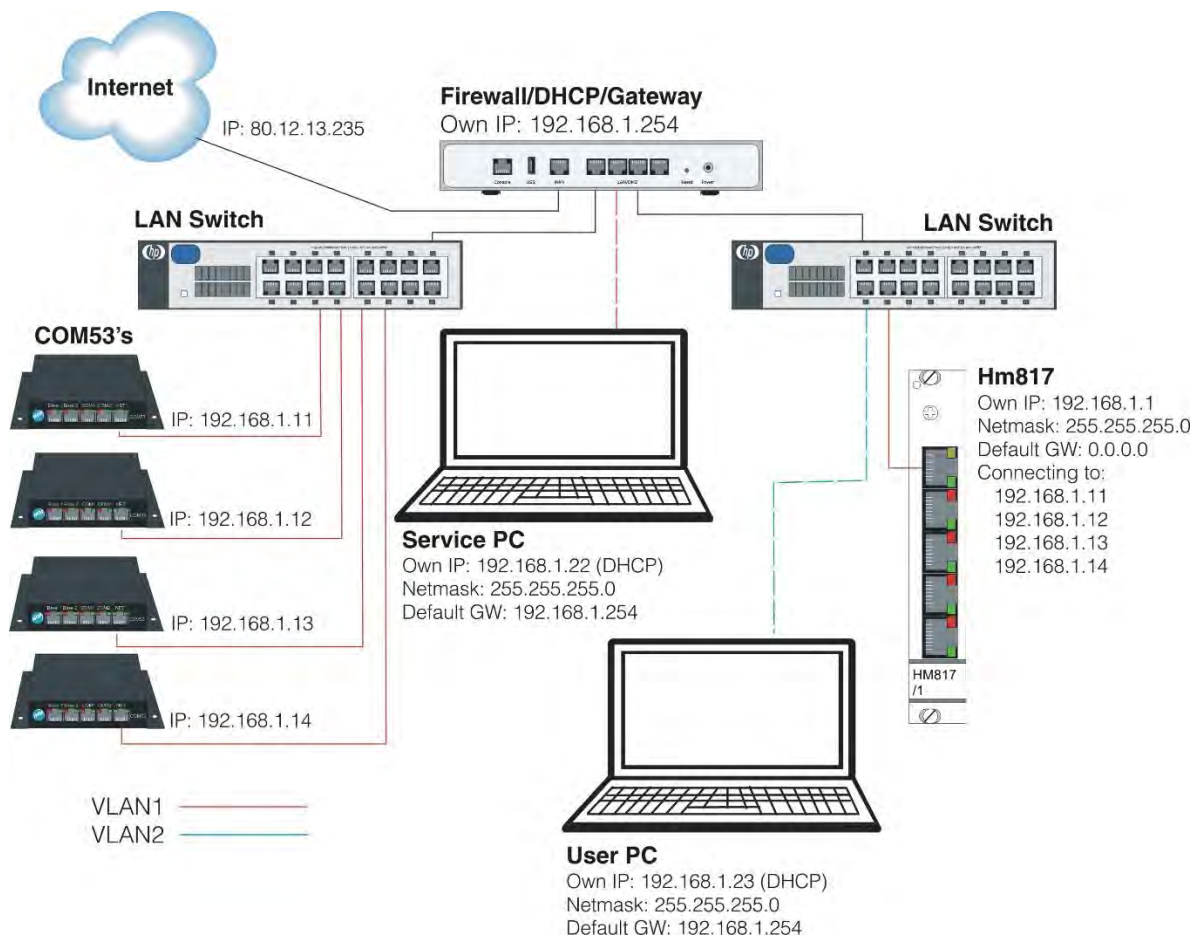
Deploying the HM817/1 and COM53 in an existing network.



In this configuration, the internal LAN is used for communications between HM817/1 and COM53. All units are coded with a Static IP-address. In this configuration, all PC's within the network can get in contact with the IHM modules. Depending on the Firewall setting it may be possible to access the Internet from any PC in the network.



5.2.3 Internal LAN with VLAN



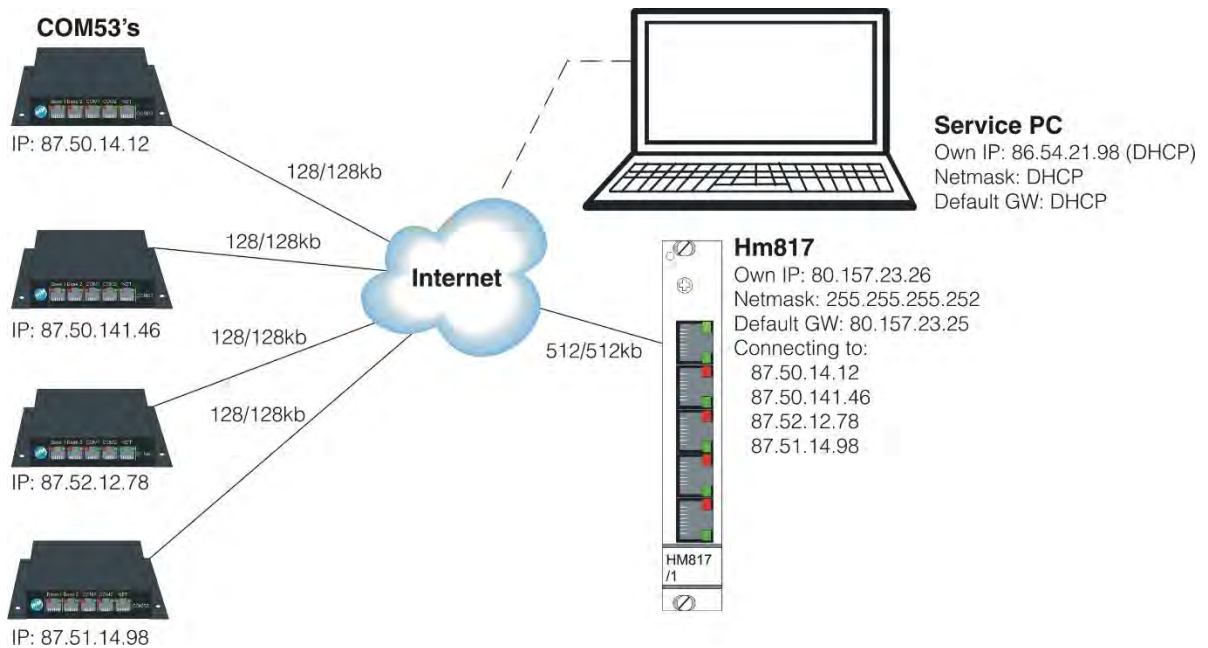
In this configuration, the internal LAN is used for communications between HM817/1 and COM53. All IHM units are coded with a static IP-address, but at the same time are all the ports in the switches that are used by IHM modules defined as a VLAN (Virtual LAN). Meaning that all communications via these ports only are switched to one of the other ports within this VLAN. In this configuration, the proprietary IHM communication is therefore limited to these ports and can therefore not communicate with other ports.



5.3 WAN

In the below described examples we use network connected over more sites. The connections between networks are provided via the Internet. The characteristic of this kind of network is that the IP-Space does not use the same IP-addresses in both ends of the network.

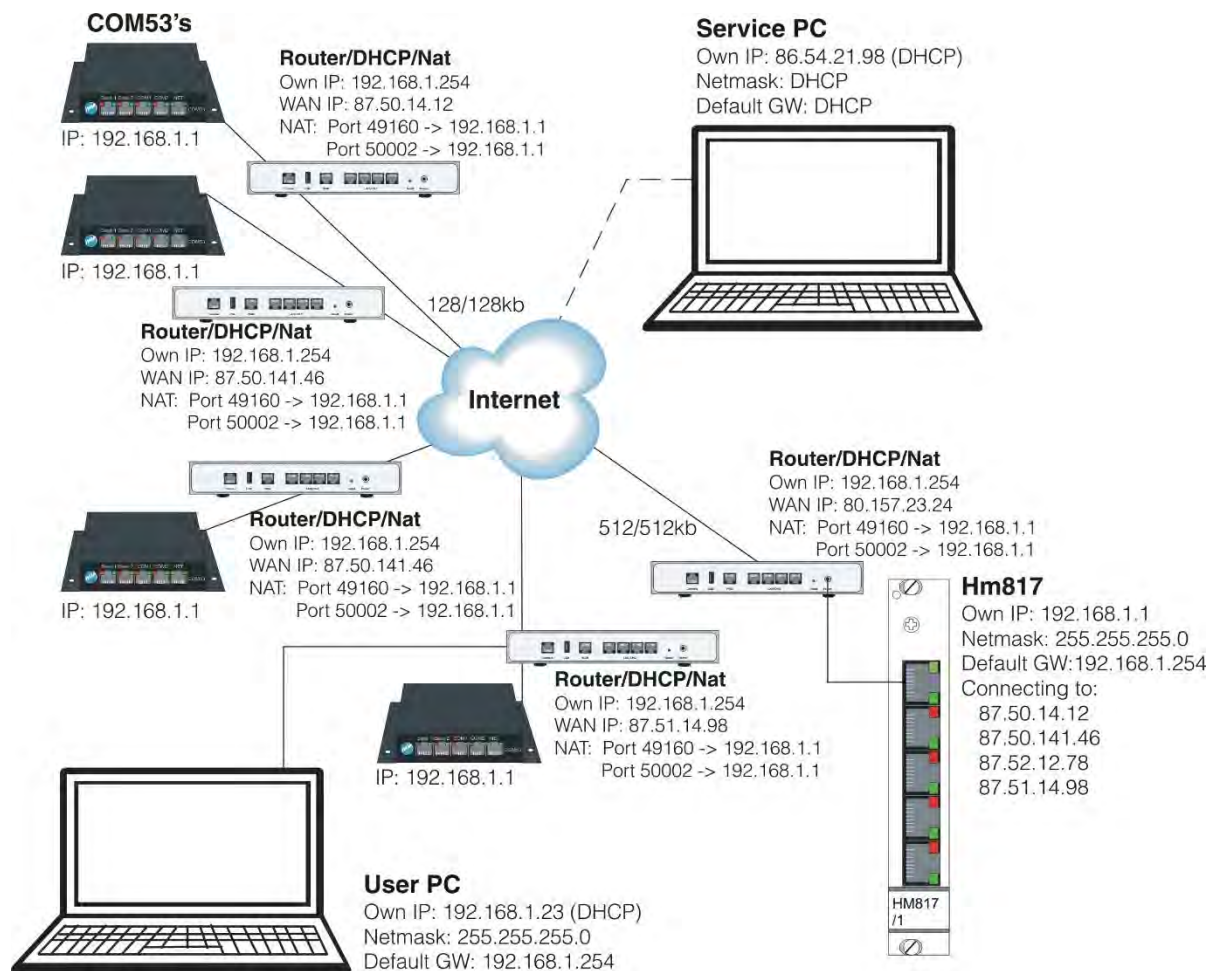
5.3.1 Internet direct connection



This system configuration is the simplest possible and does not require special configuration of the network. However please note that the Internet connection must be configured with **static IP**. This can be defined upon ordering from your ISP. For Service purposes, you can access the IHM modules from any Internet PC with the correct IHM software installed.



5.3.2 Internet via Routers

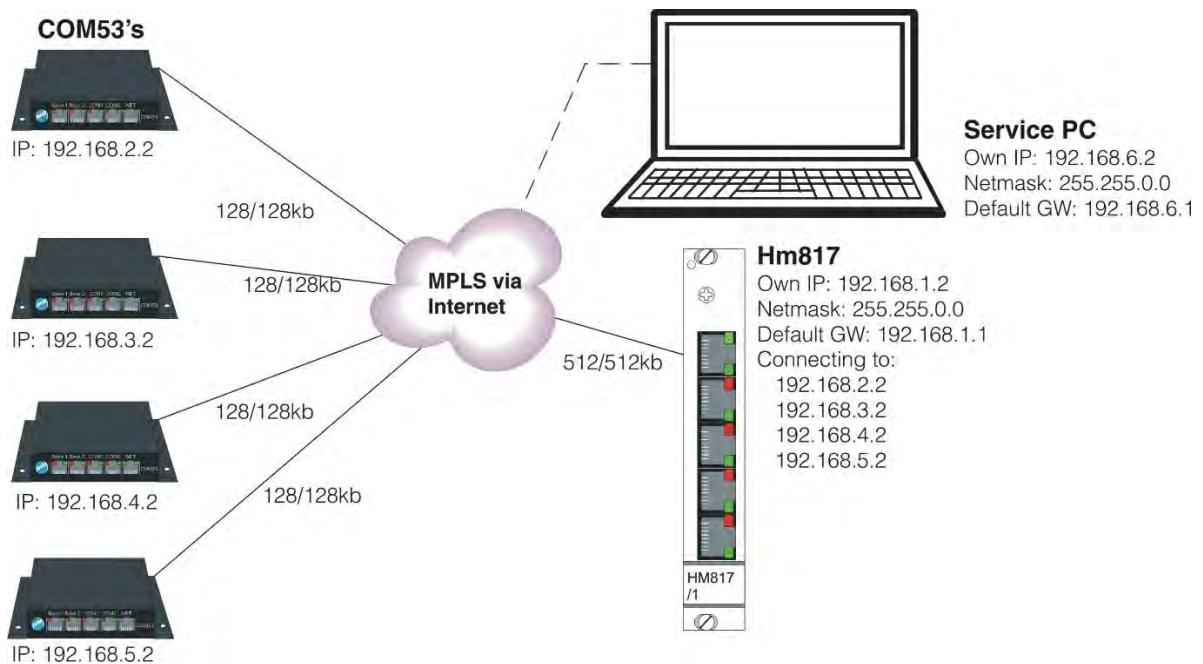


In the above system configuration, we use a router at each Internet access point. The IHM modules are connected at the LAN side, whereas the Internet is connected to the WAN side of the router. In this configuration, we can on the WAN side either use Fixed IP or Static IP addressing, as it is the router who handles the DHCP request towards the ISP. But still it must be fixed/static IP addresses. In this configuration, you can still connect user PC's to a router, if you remember the required speed/bandwidth for the VoIP – 128kB/s per voice channel. In some routers, this may be possible to configure. It is called QoS (Quality of Service), here you may reserve bandwidth for defined services.

Using this configuration, it is possible to connect more IHM units at the same Internet access point. You just must remember to program the NAT routings in the IHM Slave modules and program more IP-addresses in the Master unit for each of the slaves on the same IP-address.



5.3.3 Internet using MPLS



In this system configuration, you buy a service from the ISP where you get a private network routed via the Internet. In this solution, we use internal IP-addresses as it would have been if it was the same network. The network is closed, meaning that you cannot get access to or from other addresses on the Internet.

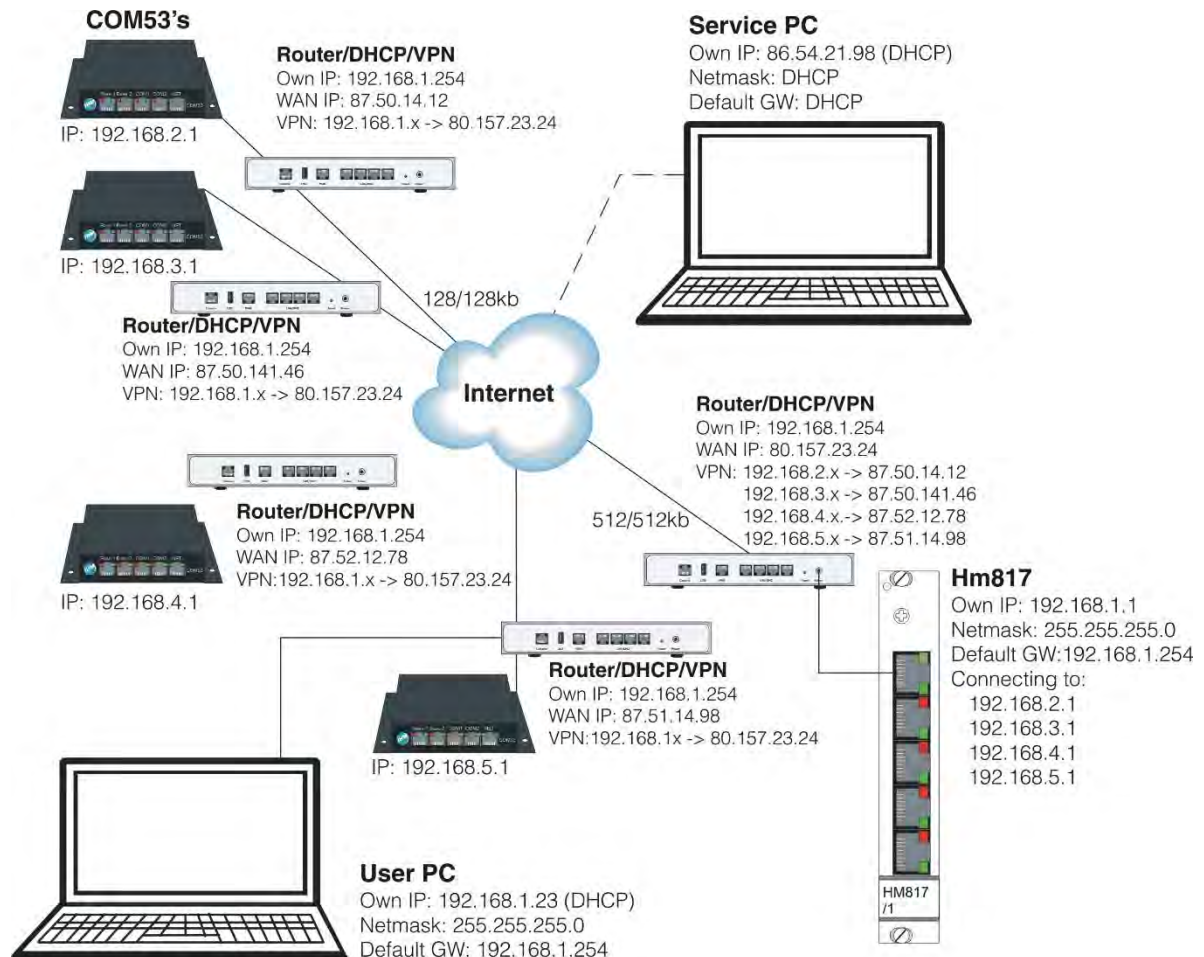
This also means that a service PC needs to be connected to a port within the MPLS network. You do not have to use external routers and there is no special configuration to be made, all this is handled by your ISP.

For this system configuration, it will be possible to have more IHM modules at the same connection, you only have to remember to order more internal IP-addresses when ordering the MPLS network.



5.3.4 Internet using VPN

This type of network hold the same functions as the MPLS described above, the difference is that you are responsible for the routers.



For this solution, a special type of routers is used. The routers hold a technology called VPN (Virtual Private Network). In these routers, you configure some rules about where to find the "internal" IP-addresses. When calling one of these addresses the router automatically establish a connection via the Internet to the required other router.

The communication between the two routers is encrypted.

In this type of solution, it is possible to have more IHM modules to one connection, you only must define the IP-address space per connection instead of specific addressing.

5.4 Combining more network types

One HM817 module connects to one network type only, and there is no network connection between two or more HM817 modules.

This means that both LAN and WAN can be connected to the same IHM communications switch, not introducing any security risk at all.