TaitNet P25 network

# Installation Guide

This manual is intended for use by
Tait-approved system installers

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*Note: Technical specifications are subject to change without notice.*

# Preface

This Installation Guide aims to provide assistance to Tait-approved system installers in the work of installing and commissioning TaitNet P25 networks. Others may find the information in the guide useful for example in determining the requirements for a TaitNet P25 network. Those who are configuring routers and switches should be Cisco-certified network administrators or equivalent.

The Installation Guide gives an overview of the main tasks involved in installing a network and then detailed information about particular aspects. It assumes that you are familiar with the other documentation on the TB9100 product CD.

## Enquiries and Comments

If you have any enquiries regarding this manual or any comments, suggestions or notifications of errors, please contact Tait Technical Support.

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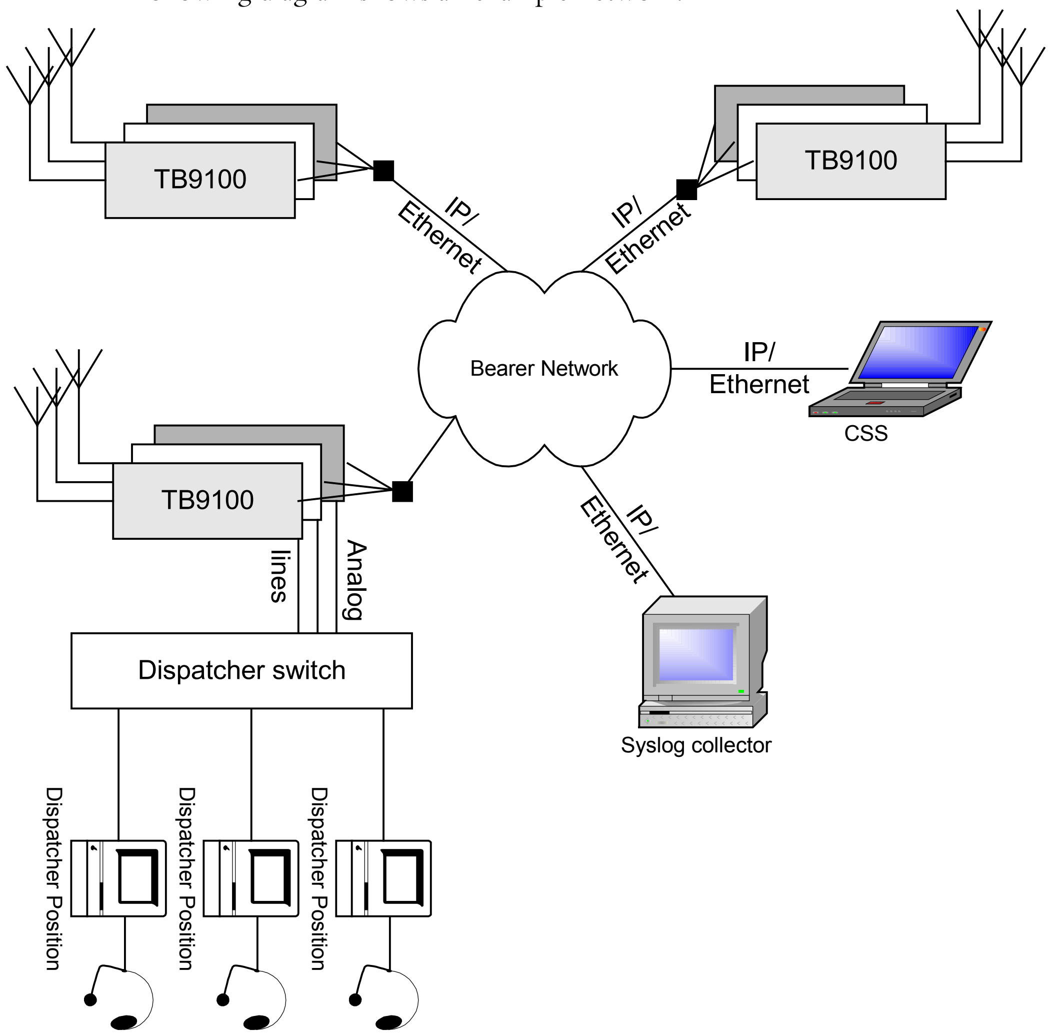
| MBA-00018-01 | May 2005 | First release. |
| MBA-00018-02 | August 2005 | Two network types introduced: Switched network and routed network. Sections added for Network Topology, Making the Transition to Digital, Cross-mode connection. Details added to linking bandwidth. |
| | | |

# Introduction

A TaitNet P25 network is a set of interconnected TB9100 base station transceivers. For an introduction to the way it functions and to the different topologies it can have, see the white paper “TaitNet P25 Networks.” The following diagram shows an example network.



# Installation Overview

The following provides an overview of the tasks to be carried out when building up a TaitNet P25 network.

Before installation begins, you should have received detailed information about the setup and configuration options that the customer wants. There should be an agreed plan for channels, talkgroups, IP addressing and many similar matters. The Equipment Setup Guide aims to provide assistance in making these plans.

# Making the Transition to Digital

In many cases, installing a TaitNet P25 network involves making a transition from an existing analog system to the new P25 system.  Here is an example plan for managing that transition.

1. Add P25 radios to the existing fleet. These radios are configured for dual mode receive and analog transmit. Users can operate them as part of the existing analog system.
2. While the existing system continues to operate, install Tait TB9100 base stations.
3. Test and commission the first channel group of TB9100 base stations, using test frequencies.
4. Cut over to the TB9100 using the existing analog frequencies. Users can be asked to use another channel on the existing system during the brief outage involved.
5. Make some test calls and then begin live operation in analog mode using Tait P25 radios and base stations.
6. After some time of analog operation, ask users to switch their radio to a P25 transmit channel or zone, with dual-mode receive.
7. Repeat steps 2-6 for each channel group until the entire system is replaced by the TaitNet P25 network.
8. Some time later, you can turn off dual mode operation in the radios and base stations. Digital only operation can improve system performance.

# Base Station Run-Up

Before installing on site, run each base station up on the bench, configuring and checking it.

## Calibrate the Reciter

If the base station is not calibrated for the switching range (lock band) that the network requires, you need carry out the following to adjust the frequency setup.

- Adjust the receiver lock band
- Tune the receiver front end
- Adjust the exciter lock band

- Set the RSSI

You need to remove the reciter from the subrack and connect it (together with a PC running the calibration software) to a Calibration and Test Unit. For details on the setup and on these procedures, see the calibration software manual or online Help. After calibration, reset the base station to take it out of calibration mode. It is now operational but still in standby mode.

## Connect the CSS

Follow the instructions in the Installation and Operation Manual to connect the CSS for the first time to the base station, to change its IP address. The IP address given in the factory is:

192.168.1.2

Make sure that your computer is in the same subnet. If you still can't connect, this may be because the base station has a different IP address. Find out its address as follows:

1. Connect your PC to the 9-pin serial connector on the back of the reciter.
2. Run a program such as HyperTerminal, Teraterm or minicom.
3. Select the following port settings: 57600 baud, 8 bits, no parity, 1 stop bit, no flow control.
4. Press the 'Enter' key. A login prompt will appear displaying the base station's IP address.

If you still have trouble connecting, consult the topic "Troubleshooting Connection Problems" in the CSS Help or manual.

Give the base station the necessary settings. Make sure you do the following:

- Set the analog gating (recommended settings: SINAD 13 dB with a 6 dB hysteresis)
- Give any digital P25 channels the correct NAC.

Put the base station in Run mode when you are finished.

## Check the Alignment

Using RF test gear, re-check the following items on the factory test sheet. This helps verify that the base station is operating correctly.

- Frequency
- Sensitivity
- Line Levels
- Deviation
- Distortion

Additional operational tests are in the Installation and Operation Manual.

## Use Test Radios

Program up radios so that they can operate using the base station, and then carry out the following functional tests:

- Group call

If the network will use supplementary services, test the following:

- Call alert
- Status call
- Message call

# Analog Line Run-Up

## Connect the Analog Line to the Dispatcher System

See Analog Line on page 25.

## Functional Test of Dispatcher

1. At the dispatch location, connect a CSS to the base station.
2. Select Diagnose > Line Interfaces > Analog Line and run a transmit test on the base station's analog line. Use a tone frequency of 1000 Hz and a level of -10 dBm.
3. Verify that a tone can be heard at the dispatch console speaker.
4. Select Monitor > Interfaces > Analog Line.
5. Use a tone generator with frequency 400 Hz and level -10 dBm or whistle into the dispatch microphone with the PTT pressed.
6. Verify in the Analog Line form that the measured RMS input level is near to -10 dBm.
7. If the base station is keyed by E & M, verify that the E-wire (input) LED is green and the Channel seize LED is green.
8. If the base station is keyed by Tone Remote and does not support MDC1200 signaling, verify that the LLGT detected LED is green, the channel seize LED is green and the overload LED is gray.
9. If the base station is keyed by Tone Remote and does support MDC1200 signaling, verify that the LLGT detected LED is green, the Received LED is green, the Channel seize LED is green and the Overload LED is gray.

### Troubleshooting

- If the overload LED is RED, the dispatch transmit HLGT level is too high or the analog line input gain is too high.
- For tone remote keying, if the LLGT LED is not green, the HLGT level is too low or the LLGT level is too low or the analog line input gain is too low.
- The MDC1200 LED will not turn green if the HLGT level is too low.

# Site Run-Up

## Set up and Configure the Switch

See Switch Configuration on page 18

## Set up and Configure the Router

You need to configure each port. You may need to set up a secondary subnet for CSS connections, a firewall with access list, and remote access from Tait for support. If the network is multi-site, you need to configure the routers for quality of service. For a routed network, see Router Configuration on page18 and for a switched network, see Router configuration on page 25.

## Intersite Functional Testing

You need to carry out the following to verify intersite operation:

Make calls from a radio at one base station to another radio at a different base station. This tests correct channel group operation.

# Network Requirements

# Network Topology

TaitNet P25 networks normally use a star topology. This means that there is a maximum of two hops between any base stations in a channel group. Multiple physical links can be provided between routers or switches to provide redundancy. It is also possible to have a second central router in position, to remove the single point of failure.

Non-star configurations are acceptable for switched networks. Although there are more links, the latency on each link is much lower, so that the absolute jitter is still low. Regardless of the physical topology, there is a virtual tree.

Non-star topologies such as mesh and ring are not recommended for routed networks. The increased number of hops increases latency and jitter. These topologies offer alternative routes, which may result in packets being received out of order. Base stations drop these packets and substitute empty values for them.

# Recommended Products

The following lists recommended network devices. It is possible by agreement with Tait to deviate from these recommendations, for example if you want to integrate the TaitNet P25 network within an existing IP network or you want to use other router types. It may be necessary for Tait to provide consultation services in order to achieve the necessary QoS requirements.

## Hubs/Switches

Cisco Catalyst 2950 Switch. You need IOC version 12.1(19) and C2940.

The standard router includes a small hub, however a separate hub or switch is needed if the number of base stations at the site exceeds the number of ports in the site router.

## Routers

If the linking infrastructure is only going to be used for remote maintenance, any suitable router can be used. Otherwise the following operating system versions, feature sets and router models are recommended.

### Feature Set

Routers need to be equipped with certain features that enable the RTP voice stream to be transported across the linking infrastructure with a minimum of delay:

- Network-based Application Recognition (NBAR): assists with the prioritization of the voice stream.
- IGMP v3: This enables routers to recover quickly from link loss. IGMP snooping in the switches must be turned off.
- The central router must be configured as a rendez-vous point (RP).

In addition, routers should support the Proxy ARP protocol. Base stations use this protocol to find out which host on the LAN will forward packets destined for the network. If different routers are used that don't support this protocol, it is necessary to set a gateway address in the base station. Currently, this is not configurable using the CSS and you must telnet into the base station and set the value.

### Standard Router

Cisco 1760 router. This router is rack mountable and can accept two wide area interface cards.

Use this router throughout the network if possible, to simplify support and logistics.

You need IOC version 12.3 (12) and C1700-IPBASE-M.

Order a version with the necessary interfaces to the digital links that will be provided (for example V.35, X.21, RS232, E1/T1 G.703 or G.704). If only analog links are available this router is used together with the modem (see below).

Firewall functions and VPN (virtual private network) support are not part of the standard router software. They must be ordered as optional software extras.

Note: The Cisco 1760 Router auxiliary port cannot be used as a WAN link. Because PPP multilink is not supported on the auxiliary port of these routers, you should not use the auxiliary port as a WAN interface for the VoIP stream. Use a WAN interface card instead.

### Large Router

The Cisco 2610 router.

Used as the core router when the standard router does not have enough physical links for the network topology.

You need IOC version 12.3 (6a or 6b) and C2600-IPBASE-M.

## Modem

Westermo 3178-0090-TD-32 DC EU

(Tait part number T1541-OM-0100)

Needed when only analog links are available.

## Console system

Contact Tait for a list of recommended console systems or to determine the compatibility of your current system.

# Linking bandwidth

The minimum bandwidth required for the bearer links between base stations depends on the type of network and on the type of calls that the base stations handle. Compressed RTP can be used where there are few base stations. It reduces linking requirements at the expense of router CPU loading.

The following guidelines can assist in determining the minimum bandwidth.

| Network Type | Call Type | Bandwidth allocation per Base Station (kbit/s) | Notes |
|---|---|---|---|
| Routed network with compressed RTP | Digital P25 | 19.2 | These values include the 25% of the bandwidth that routers require to be left for traffic other than VoIP. |
| | Analog FM | 94 | |
| Routed network without compressed RTP | Digital P25 | 38 | Compressed RTP cannot be used with Cisco routers and E1/T1 links. 25% of the bandwidth must be left for traffic other than VoIP. |
| | Analog FM | 114 | |
| Switched Network | Digital P25 | 50 | Provide at least five times the calculated bandwidth. This keeps the utilization of the Ethernet network low enough to avoid excessive packet collisions which would cause the voice quality. It is necessary because there is no QoS. |
| | Analog FM | 110 | |

When calculating the bandwidth requirement, it is not the total number of base stations that count, but the number of base stations that need to use the link in any one direction. For example, a two-site system with one base station at each site only needs bandwidth for one base station. A three-site system with two base stations at each site needs bandwidth for two base stations.

**Note**: If the base stations in the network have the repeat function disabled but do not have centralized voting (not supported in the current release), channel groups have two voice streams instead of one. The above minimum bandwidth requirements should be doubled.

If the bandwidth is >56 kbit/s, the maximum fragment delay can be reduced to 10 ms from the usual 20 ms (use the command:

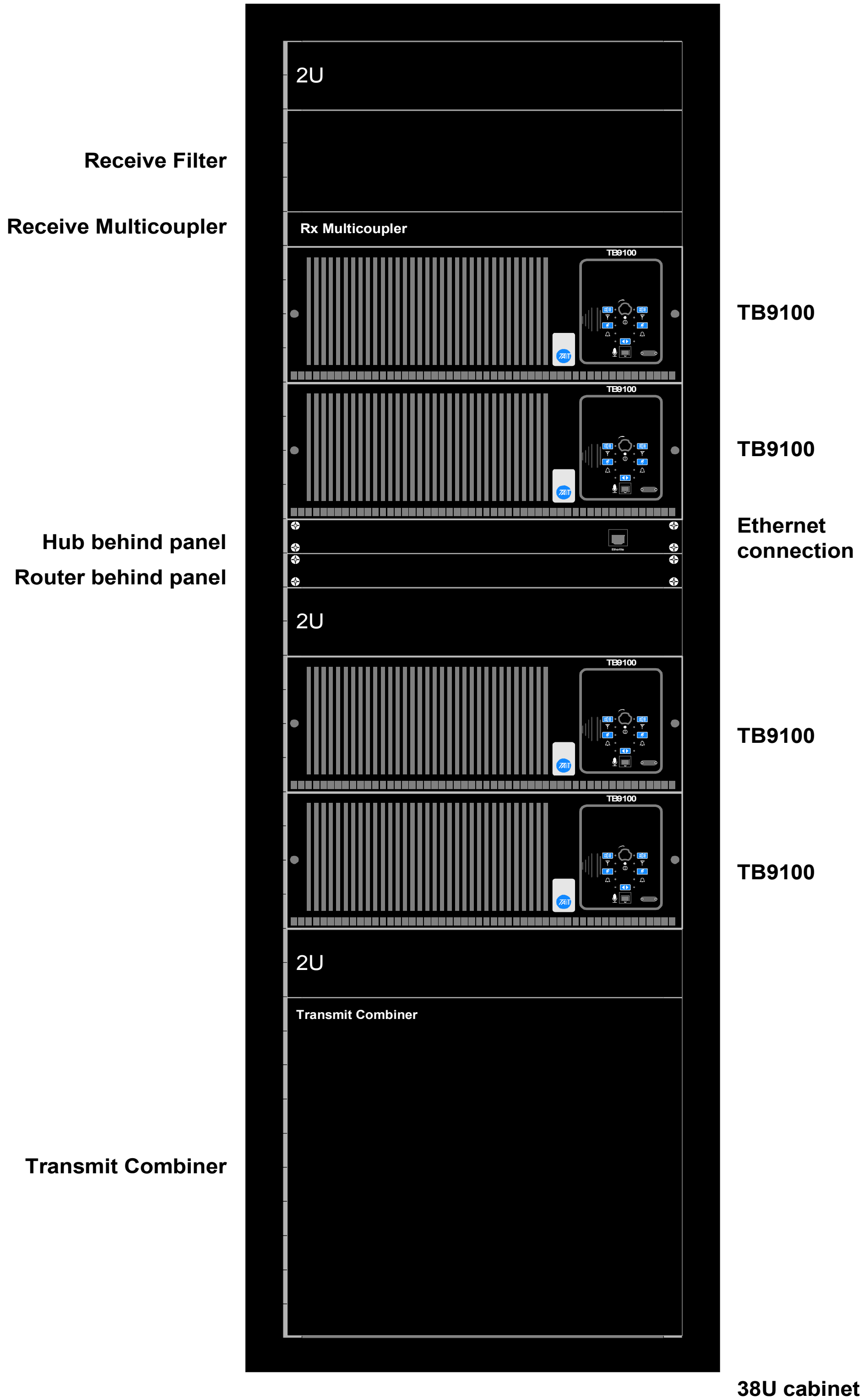
```
ppp multilink fragment-delay 10
```

If the bandwidth is >800 kbit/s, the fragment interval can be turned off.

If only serial links are available, the necessary bandwidth can be obtained by combining several links with modems. The PPP protocol handles the multiple links.

# Cabinet or Rack Layout

A standard 38U cabinet can be used to house four TB9100 base stations complete with antenna multicoupling hardware.




Other rack layouts are possible, however adequate ventilation of the cabinet is required to provide cooling for the TB9100 base stations. A 2U space must be provided between the top of the cabinet and the first TB9100 to provide an airflow path out the top of the cabinet. For further information on ventilation for the TB9100 refer to the section on cabinet and rack ventilation in the TB9100 Installation and Operation Manual.

Associated with the TB9100 base stations are a router and a switch. These are mounted in the rear of the cabinet, for ease of connection to the TB9100 base stations. The Ethernet port on the rear of each TB9100 is connected to a port on the switch.

The router is required to provide remote access to the TB9100 and is connected to a port on the switch.

An adapter front panel (Tait part number TBA60C0) can be used to provide easy connection of a CSS to any one of the TB9100s in the cabinet for programming and configuration. The adapter front panel is 1 U high and has an RJ45 Ethernet port. This port is connected via an Ethernet cable to the switch. If there is only one base station in the cabinet, the port is connected directly to the base station digital line via an Ethernet crossover cable.

# Network Configuration

## Standard IP Addressing Scheme

The addressing scheme uses RFC1918 to implement private IP address classes. These can be easily summarized for static routing throughout the network.

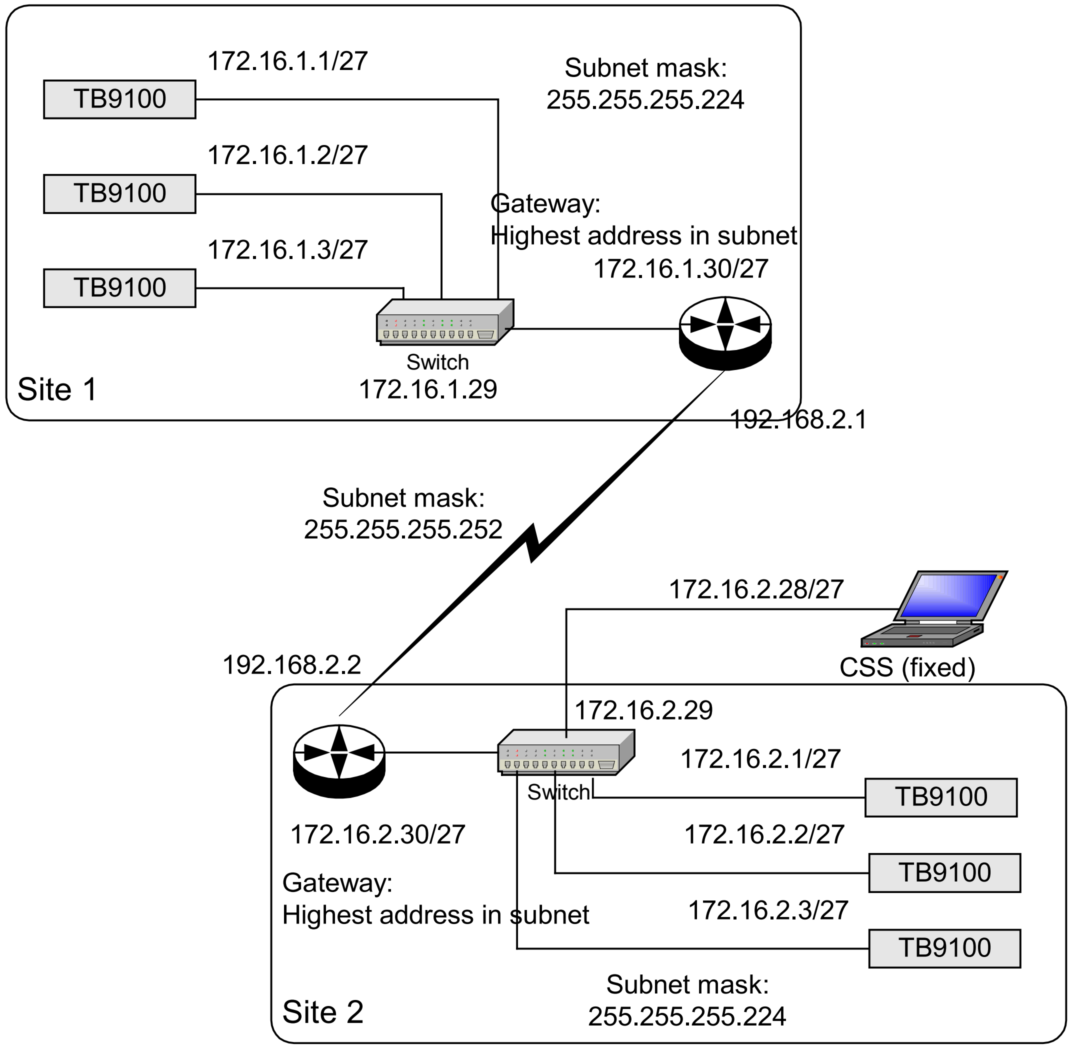
The addresses for loopback and for core links use the 192.168.x.x. scheme with 255.255.255.252 as the subnet mask. This allows for four addresses on each link: one to describe the network, one for broadcasting to all devices on this link, and two to describe each end of the point-to-point link.

The addresses for the Ethernet links on the far end of the routers use the 172.16-31.x.x network structure with 255.255.255.224 as the subnet mask. This allows up to 27 base stations on each LAN segment.

Each router in the network is configured with a loopback address.

The following shows an example addressing scheme for a two-site network.




## Parameter Values

### Equipment on Site LANs

Devices on a site LAN are given an IP address based on the following scheme: 172.W.X.Y/27

Netmask: 255.255.255.224

Where:

W = 16-32. Adjacent customers use different values. If you run out of W values, use different numbers for the three MSB of Y.

X = site number

Y = device number. 30 = router, 29 = switch loopback address. 1+ = base station (number from 1 up). 28- = CSS PC (number from 28 down).

### WAN Links

The routers are given an IP address for their core link using the following scheme:

192.168.2-255.X/30

Where X = site number.

In addition each router needs a loopback number:

192.168.0.X/32

The gateway address for each site (router) is the highest address in the subnet.

### UDP Port Numbers

Tait uses the following port numbers for communications between elements on the network.

- Syslog messages from base station to syslog collector: 514
- Tait APCO RTP: 27260
- Tait APCO RTCP: 27261
- Supplementary services protocol: 27000
- CSS to Network board protocol: 27100
- Base station to controller trunking messages (not yet implemented): 27258

### Multicast Addresses for Channel Groups

Multi-cast addresses are in the range 224.0.0.0 through 239.255.255.255.

- Do not use an address between 224.0.0.0 and 224.0.0.255. These addresses are reserved.
- It is best to use a contiguous range of addresses, in part because this makes it easier to program routers.

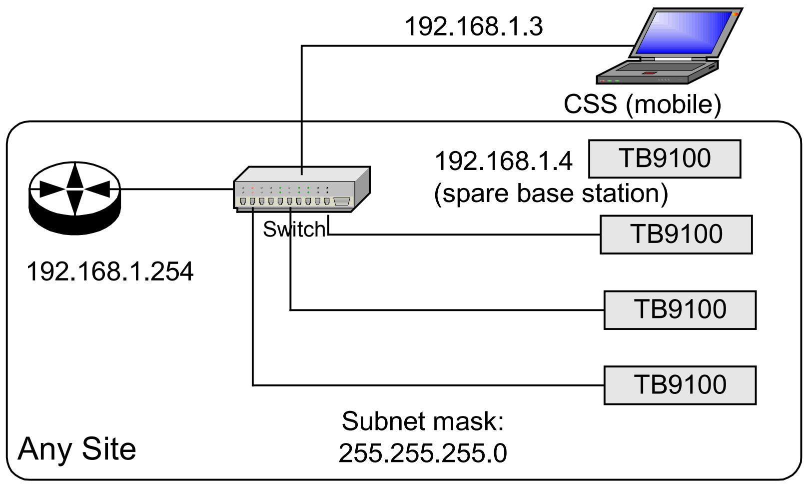
## Address Tables

You can use the web calculator at http://jodies.de/ipcalc to generate tables of the addresses that belong to any subnet.

# Laptop subnet

Connecting the CSS to a switch port at a site is made easier if each site router is programmed with a secondary subnet. Each site uses the same secondary subnet. The laptop's IP address and subnet then do not need to be changed when the technician connects to a base station in a different site. When using the laptop subnet, the CSS can only connect locally to base stations on the same site.

The following shows example addresses for a laptop subnet that supports one CSS and one spare base station.




## Parameter Values

192.168.1.0/24 Secondary subnet

255.255.255.0 netmask of the secondary subnet

192.168.1.1 Windows default

192.168.1.2 Base station default IP address ex factory

192.168.1.254 Secondary address of each router's Ethernet port. Also the default gateway for all hosts in this subnet.

192.168.1.3-253 Address of laptops and spare base stations.

# Firewall considerations

Where a TaitNet P25 network is connected to an organization's intranet or to the public internet, a firewall is needed to protect the TaitNet P25 network from unauthorized access and to prevent the voice stream from inadvertently being multicast beyond the TaitNet network. The default option is to use the firewall facilities of the central router.

Firewall needs to let the following through:

- CSS communications to/from base stations
- Base station communications to the syslog collector

# Remote access and support from Tait

To provide the best level of support, it is helpful if Tait engineers can remotely access the TaitNet P25 network. A TCP/IP connection is required and this enables Tait to monitor and carry out diagnostic tests on any TB9100 base station on the network. The base station firmware can also be remotely upgraded. There are several common remote access methods:

1) VPN via Internet

2) Existing Remote Access Solution that the customer uses

3) Dial up modem/router

Tait prefers method 1, as it is very reliable and secure when correctly installed. Method 2 may be a sensible choice if the customer already has a remote access solution in use, and is willing to allocate Tait Technical Support a login. The least favored option is dial up, due to poor reliability and performance.

# Routed Network

Here are some guidelines for configuring a linking infrastructure that is based on switches and routers. In this design, the TaitNet P25 network is essentially a wide area network and routers provide the protocol conversion between the fast LAN links at the site and the slower wide-area links.

## Switch Configuration

The Cisco switch MUST have no ip igmp snooping in its configuration because it doesn't understand version 3 igmp. Multicasting just doesn't work if you have it on. Cisco Catalyst 2950 switches have **ip igmp snooping** on by default. This needs to be disabled.

Disable it by:

```
conf t
no ip igmp snooping
```

## Router Configuration

Routers in the network will need to implement the procedures required by the multicast routing protocol to join and send to multicast groups. The core router is the Rendezvous Point (RP) router.

The only traffic control facilities are a simple priority scheme, to give RTP voice traffic priority over other traffic.

### Quality of Service

Routers must be configured for quality of service. This gives the voice stream (RTP packets) priority over other data on the linking infrastructure.

The following set of commands provide the standard router configuration for quality of service:

```
access-list 101 remark BS RF Sourced Traffic
access-list 101 permit ip any any dscp ef

class-map match-all Gold
 match access-group 101

policy-map WanPolicy
 class Gold
   priority percent 75
 class class-default
  fair-queue 16

interface Multilink1
 service-policy output WanPolicy
 ppp multilink
 ppp multilink fragment-delay 20
 ppp multilink interleave
 multilink-group 1
 ip rtp header-compression iphc-format
 ip tcp header-compression iphc-format
 no ip mroute-cache
 no ip route-cache
 load-interval 30
```

In addition, site (non-core) routers need the following:

```
int eth0
  ip igmp version 3
```

## Explanations

### access-list 101

This matches any udp packets marked with the DSCP code for EF (Expedited Forwarding, 101110 binary). The Base Station marks all RTP voice traffic as EF.

### class-map Gold

This defines a class of traffic.

### policy-map WanPolicy

This creates two classes of traffic for the routers to handle:

- Gold (RTP Voice traffic), which has priority access to 75% of the bandwidth
- Default, which has best-efforts on everything else and 16 queue slots (Cisco default)

NB. Cisco recommends that you only reserve up to 75% of the link bandwidth and leave 25% for best-efforts (eg Router protocols). For priority allocations, you cannot specify above 75% unless you add some additional configuration to reset the limit (Google says look here http://www.cisco.com/en/US/products/sw/iosswrel/ps1839/products_feature_guide09186a0080087af0.html and here http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122t/122t2/ftllqpct.htm#38308)

A priority allocation sets up a fast track through the router for matching packets. It also drops packets if they exceed the reserved bandwidth and there is congestion.

### interface Multilink1

This defines a PPP multilink bundle so that we can allocate the Quality of service policy to an interface and also make use of Link Fragmentation and interleaving. This is necessary for links slower than 600 kbps. Above this limit it takes less than 20 milliseconds to transmit a maximum (1500 byte) packet so the `ppp multilink fragment-delay 20` rule never applies. If the link speed is 56 kbps or above (T1), it is worth changing this to `fragment-delay 10` to give tighter jitter control (better voice quality). You can't use 10 ms on really slow links because the resulting packets would only just manage to contain the IP headers.

### no ip mroute-cache and no ip route-cache

These are really important. NB. the ip route-cache command doesn't apply on 800 routers (edges). They stop the router optimising the handling of packets and bypassing the rtp compression config.

### load-interval 30

This is nice to have because it makes the monitoring of data on the interface work. The default is 5 minutes and Cisco says Average number of bits and packets transmitted per second in the last 5 minutes.

The 5-minute input and output rates should be used only as an approximation of traffic per second during a given 5-minute period. These rates are exponentially weighted averages with a time constant of 5 minutes. A period of four time constants must pass before the average will be within two percent of the instantaneous rate of a uniform stream of traffic over that period.

Using a 30 second period and a long voice call should allow you to check the priority offered rate during say a one minute call.

### ip igmp version 3

This makes the site routers use IGMP version 3 over the Ethernet interface to determine whether any multicast groups are in operation. The default is IGMP version 2. Most documents tell you that Version 3 adds site-specific multicast ie. You can use it to request multicast only from certain sources. This is true, but we don't need that feature. Version 3 also adds a periodic general query from the router. All multicast-enabled hosts on the sub-net should respond to the general query with all current multicast memberships (suppressing any that are the same as responses from other hosts). The advantage of this addition to the protocol is that router loss and recovery (eg power cycling), switch loss and recovery (power cycling or cables disconnections and reconnections) do not cause the loss of multicast membership information.

On Version 1 and Version 2 the router remains a group member until it resets or it times-out waiting for a membership response to a membership query. The router only sends queries for groups that it is currently a member of. If a Base Station is a group member but it misses the query for any reason, the router will delete that group and will not update information about the Base Station until it joins a new group.

### Bandwith

You must remember to set the bandwidth on the physical interfaces making up the multilink bundle, for example:

```
bandwidth 19
clock rate 19200
```

This is so that the service policy percentage has something to work on (the clock rate doesn't get passed to the parent).

### Compressed RTP

Routers on T1 links can be configured to use compressed RTP if the bandwidth of the links is at or below 1.4 Mbps. For T1 links with a bandwidth above 1.4 Mbps, compressed RTP cannot used.

### Enabling the ICMP Router Discovery Protocol

Cisco says this is enabled by default, however it is not. Configure it for each Ethernet interface using the command:

```
ip irdp
```

This allows the base stations connected to that Ethernet interface to find the IP address of the router, so that they can check that the router is Tait-approved.

### Configuring the router as 'Tait approved'

To prevent the base stations from generating regular syslog messages indicating that a router is not Tait-approved, you need to set a message of the day, using the following command:

```
Site1(config)#banner motd c
  message
  c
```

Where *c* is an arbitrary character that signifies the start and the end of the message.

The message you set must be 'Tait approved router.' In future, you may also be required to provide a signature string provided by Tait.

For example:

```
Site1(config)#banner motd *
Tait approved router
*
```

### Checking that the QoS Configuration is Working

Once you are logged into the router and have used the Enable command to give you access.

```
 clear ip rtp header-compression
 clear counters multi1
 Clear "show interface" counters on this interface
[confirm] confirm
```

Now arrange for a Base Station to send a voice stream through the network.

```
sho ip rtp header-compression
Interface Multilink1 (compression on, IPHC)
   Rcvd:    31577 total, 31500 compressed, 0 errors, 0 status msgs
            0 dropped, 0 buffer copies, 0 buffer failures
   Sent:    28000 total, 27980 compressed, 0 status msgs, 0 not predicted
            19000 bytes saved, 35000 bytes sent
            2.5 efficiency improvement factor
   Connect: 16 rx slots, 16 tx slots,
            5 misses, 2 collisions, 0 negative cache hits, 16 free contexts
            99% hit ratio, five minute miss rate 0 misses/sec, 0 max
```

Check that it says compression on, if it doesn't you have not set RTP header-compression at both ends. Make sure that the hit ratio is in the 90% area. Make sure the efficiency improvement factor is above 2

**Note:** If you do this without clearing the counters it is typically around 1.1 because all of the other UDP traffic doesn't compress and dilutes the measurement.

Make sure that the Sent total and compressed counters are close in value.

```
sho policy-map int multi 1
Multilink1

Service-policy output: WanPolicy

  Class-map: Gold (match-all)
    31 packets, 1426 bytes
    30 second offered rate 21000 bps, drop rate 0 bps
    Match: access-group 101
    Queueing
      Strict Priority
      Output Queue: Conversation 24
      Bandwidth 75 (%)
      Bandwidth 14 (kbps) Burst 350 (Bytes)
      (pkts matched/bytes matched) 31/1426
      (total drops/bytes drops) 0/0

  Class-map: class-default (match-any)
    335 packets, 24210 bytes
    5 minute offered rate 0 bps, drop rate 0 bps
    Match: any
    Queueing
      Flow Based Fair Queueing
      Maximum Number of Hashed Queues 16
      (total queued/total drops/no-buffer drops) 0/0/0
```

Just try to check that packets matched for access-group 101 is non-zero and that total drops is very small. The packets matched will always match the number of packets but the ratio of bytes to bytes matched should be approx 1:2.33 showing that RTP compression is giving the expected benefits. Note that the bandwidth of 14 (kbps) is smaller than the measured offered rate. This is because the offered rate is determined before compression but the checking against the allocated bandwidth is after compression (at least it seems to work like that and I cannot find anything at Cisco which contradicts this interpretation. Cisco do say that the offered rate is before compression but don't say when the check against the dedicated bandwidth is performed).

```
  sho int multi1
  Multilink1 is up, line protocol is up
  Hardware is multilink group interface
  Internet address is 192.168.1.1/30
  MTU 1500 bytes, BW 19 Kbit, DLY 100000 usec,
     reliability 255/255, txload 1/255, rxload 1/255
  Encapsulation PPP, LCP Open, multilink Open
  Open: IPCP, loopback not set
  DTR is pulsed for 2 seconds on reset
  Last input 00:00:16, output never, output hang never
  Last clearing of "show interface" counters 00:37:14
  Input queue: 0/75/0/0 (size/max/drops/flushes); Total
output drops: 0
  Queueing strategy: weighted fair
  Output queue: 0/1000/64/0/0 (size/max
total/threshold/drops/interleaves)
     Conversations  0/7/16 (active/max active/max total)
     Reserved Conversations 0/0 (allocated/max allocated)
     Available Bandwidth 0 kilobits/sec
  30 second input rate 0 bits/sec, 1 packets/sec
  20 second output rate 0 bits/sec, 1 packets/sec
     2647 packets input, 184794 bytes, 0 no buffer
     Received 0 broadcasts, 0 runts, 0 giants, 0 throttles
     0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored,
0 abort
     2658 packets output, 181332 bytes, 0 underruns
     0 output errors, 0 collisions, 0 interface resets
     0 output buffer failures, 0 output buffers swapped out
     0 carrier transitions</i>
```

The important thing to check here is that `Output queue: 0/1000/64/0/0 (size/max total/threshold/drops/interleaves)` has interleaves in it.

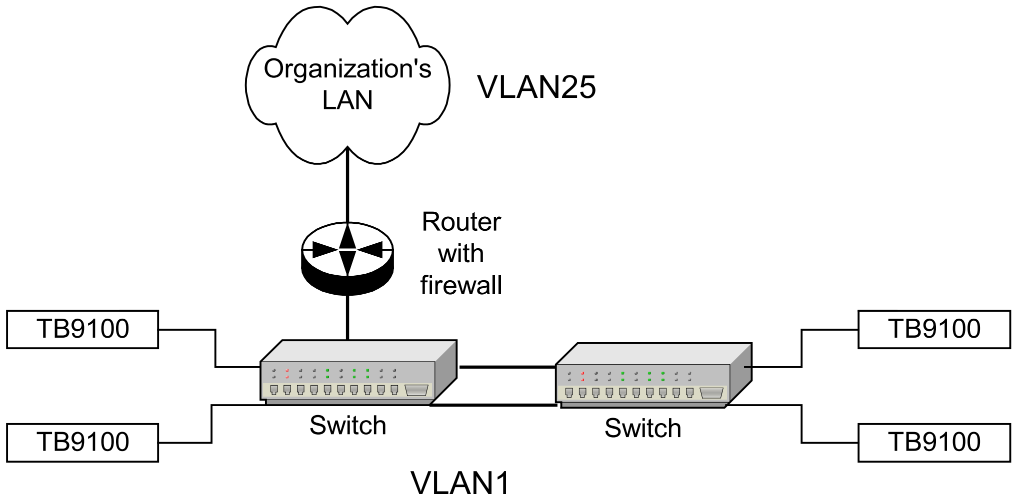
**Note:** You may have to do a read using the CSS to force the interleaves number to appear. Sometimes Cisco shows interleaves when the counter is zero, Sometimes (mostly) they don't.

# Switched Network

Here are some guidelines for configuring a linking infrastructure that uses switches. In this design, the TaitNet P25 network is essentially a LAN. Sites are linked using switches instead of routers and switches. This can be a cost-effective solution, but it needs more linking bandwidth. Each TB9100 requires 50 kbps for digital P25 calls or 110 kbps for analog FM calls. Because there is no quality of service (this can only be provided by the router), the combined bandwidth requirement of the available channels (TB9100 base stations) should not exceed 20% of the total available bandwidth. This recommendation ensures minimal jitter and becomes more important as the number of channels increases, increasing the likelihood of contention for access.

In the routerless switched LAN, use is made of virtual LANS  (VLANs). Often, the whole TaitNet P25 network is configured as a single VLAN. If the TaitNet P25 network needs to be connected to the organization's LAN, that LAN needs to be configured as a separate VLAN. In larger networks, it makes sense to set up a VLAN for each channel group.

In the following example, the organization's LAN is configured as one virtual LAN (VLAN25) and the TaitNet P25 network as another (VLAN1). A router is needed to route data between VLANs, in this case in order to provide CSS access over the organization's LAN.




## Switch configuration

For most switch ports, enable portfast.

```
configure terminal \ interface fastethernet 0/1 \ spanning-tree portfast.
```

For switch ports that are connected to another switch, do not have portfast configured. This protects the network against Ethernet loops that could cause packet flooding.

The switches also need to be set up for VLANs. Ports on the switch are naturally part of VLAN1; if you want them to be part of another VLAN then they must be configured that way. It is important to note that configuration from a telnet connect can only happen on VLAN1.

Switch ports that are connected to other switches or to the router need to be in trunked mode. This enables the port to carry data for all VLANs. All other ports are assigned to a single VLAN in access mode.

The switch port that connects to the router needs to be in trunked mode and duplex mode:

```
interface FastEthernet0/1
 description To Router
 switchport mode trunk
 no ip address
 duplex full
 speed 100
 spanning-tree portfast
```

Any ports that are connected to a host not on VLAN1 need to be configured as a part of another VLAN, for example:

```
interface FastEthernet0/2
 switchport access vlan 25
```

All ports need to have igmp snooping disabled, see Switch Configuration, on page 18.

## Router configuration

The router's Ethernet interface needs to be divided into sub-interfaces, 1 per VLAN operating on that network. Then each subinterface is configured as a different network, for example:

```
interface FastEthernet0/0
 no ip address
 speed auto
 full-duplex

! subinterface for VLAN1
interface FastEthernet0/0.1
 encapsulation dot1Q 1 native
 ip address 172.16.16.30 255.255.255.224
 ip pim sparse-dense-mode
 ip irdp
 no ip route-cache
 ip igmp version 3
 no ip mroute-cache
 no snmp trap link-status

! subinterface for VLAN25
interface FastEthernet0/0.25
 encapsulation dot1Q 25
 ip address 172.25.206.252 255.255.0.0
 no ip route-cache
 no ip mroute-cache
 no snmp trap link-status
```

# Monitoring the Ethernet LAN

Ethereal is a free-ware tool for checking what happens on an Ethernet LAN. It is available for Windows and Linux. You need a copy on a test machine (for example, the one running the CSS) if you are serious about diagnosing the network and the base station. Read the manual/help.

To configure a Cisco switch so that Ethereal can capture data for the other machines attached to the switch (BS and router), configure the switch with the following and then plug your PC into port 12 for doing Ethereal monitoring.

```
monitor session 1 source interface Fa0/1 - 11

monitor session 1 destination interface Fa0/12
```

The first line tells the switch to monitor inputs from ports 1-11. The second line tells the switch to output the monitored inputs on port 12.

**Note:** This configuration stops the PC working as an IP host. If you want to use the CSS to communicate with a base station, move the PC to another port. If you only want to run the CSS and monitor communications between the CSS and base stations, you don't need this configuration.

# Analog Line

The Tait TB9100 analog line connector provides four-wire audio and E & M connections. Make sure that you use the keyed RJ45 connector to prevent the analog line being inadvertently plugged into the TB9100's Ethernet socket.

This section contains the following:

- Typical circuits for the 4-wire audio and E & M signaling
- Protection measures for the analog line circuits
- An overview of tone remote and MDC1200 signaling

# Ordering a US Telco Line

When ordering a Telco line in the US, specify the following:

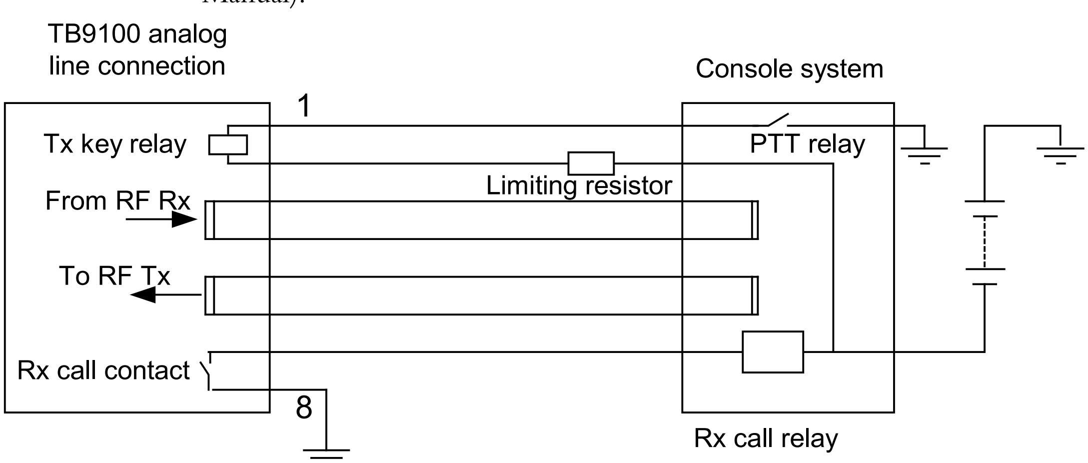
- USOC:RJ1CX. (If connecting to a wall jack socket. You must connect the analog line using a Tait cable with a RJ1CX adapter.)
- Facility Interface Code 04N02 (for a 4-wire line), or TL31E (for a 4-wire E & M line)
- Service Order Code:7.0Y.

# Local Connection to the Console System

When the TB9100 is locally connected to a console system, you can use E&M keying. The E & M input provides the capability to request the TB9100 to key the RF transmitter. The E & M output may be used to indicate that audio is present on the receiver output

The E & M circuits are implemented as solid state relay equivalents. The E & M connections are voltage free and require an external power supply if they are used.

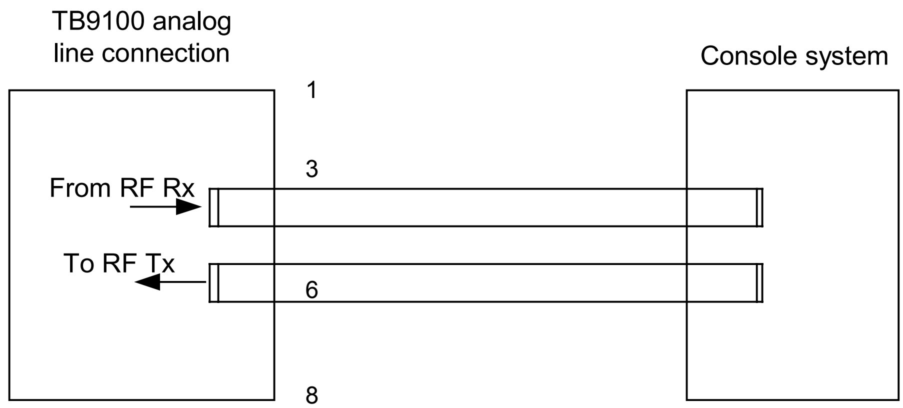
Note. The E & M input is used by the TB9100 as an input to its voting software (see the Signal Voting and Switching section in the Installation and Operation Manual).



When the console system wishes to turn on the transmitter it closes the Press to Talk (PTT) Relay. The corresponding Tx Key Relay in the TB9100 closes and transmission commences. Transformer isolated audio is modulated onto the transmitter carrier. When the PTT relay is released the transmission will cease. (Note: The TB9100 uses solid state relay equivalents). When the TB9100 receiver detects a call, the Rx Call relay closes indicating to the console system that receive audio is available on the console system's transformer-coupled input.

# Remote Connection to the Console system

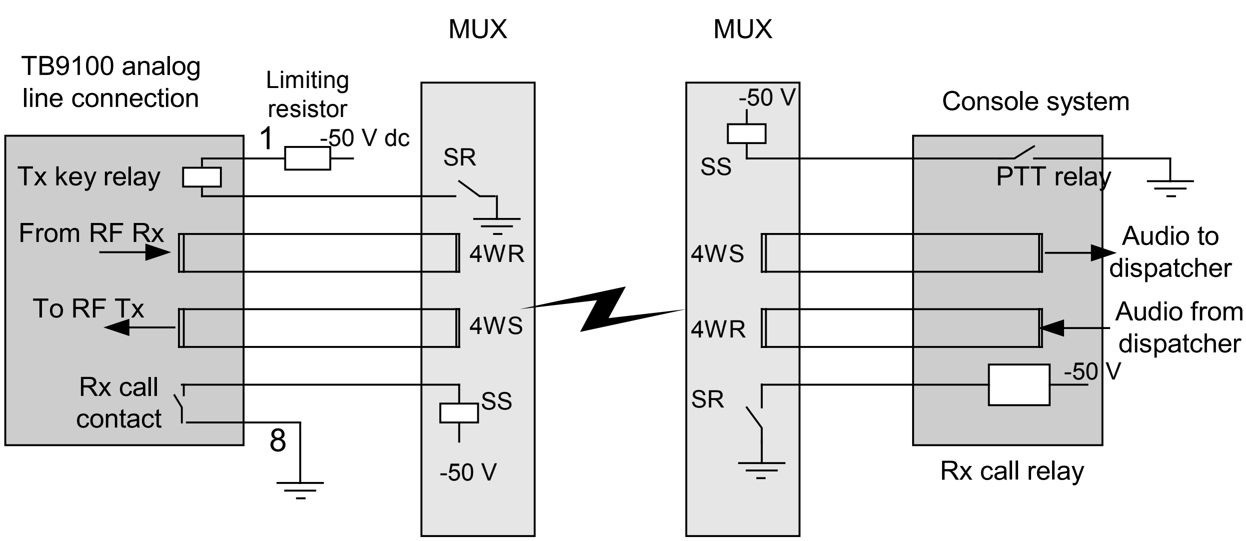
When the console system is located at a distance from the TB9100 it may be inconvenient to use E & M signaling for the TX Key and Rx Call functions. In this case a 4-wire audio-only connection may be used; the E & M connections are replaced by tone signaling.




The Console system indicates Tx Key to the TB9100 by using a low level tone: typically 2175 Hz at 20 dB below the speech level. The console system indicates Rx Call to the operators using a voice operated switch.

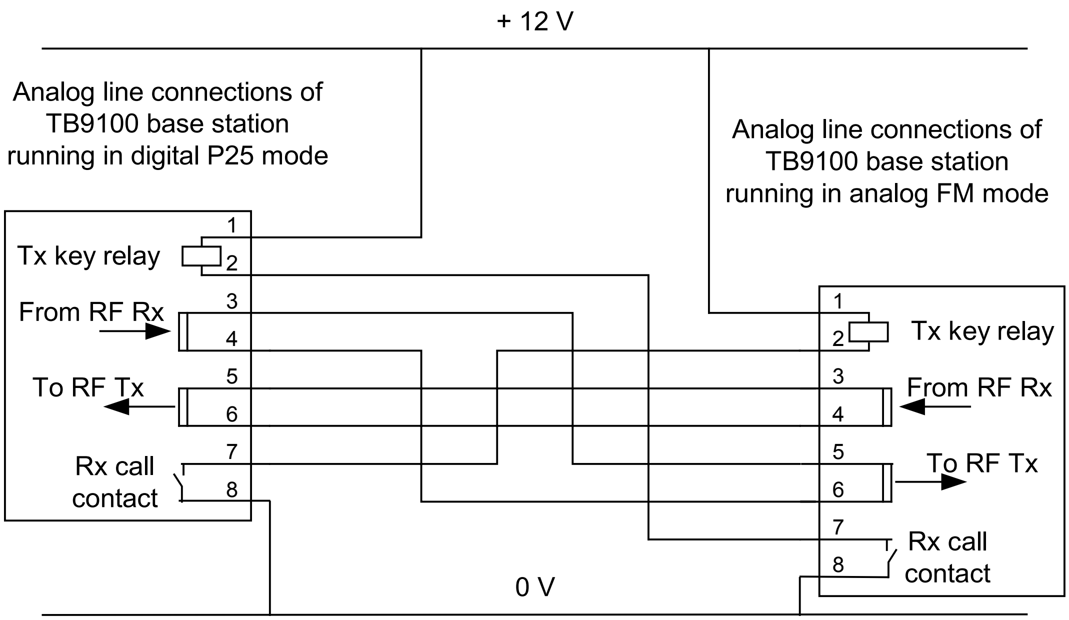
# MUX Connection to the Console System

If the remote connection uses MUXes, the E & M signaling lines can be used to control the operation of the MUXes.



# Cross-Mode Connection

The analog line of a TB9100 base station can be connected to a legacy FM base station to form a cross-mode repeater. The following circuit shows how this can be done using two TB9100 base stations. Vary this as needed to reflect the pinouts of the third party base station.




The 12 V supply can be provided by the auxiliary power output of the TB9100 PMU.

Note that in practice, if two TB9100 base stations are to be linked to form a cross-mode repeater, this would normally be done using the digital line.

# Circuit Protection

It is extremely important that the analog line is adequately protected against lightning strike and other adverse events. In this respect, it is incumbent on the installer to comply with the standards organization or regulatory body of the country of installation. While it is outside the scope of this manual to provide comprehensive information on this subject, the following advice is offered. It may or may not be compliant.

## E & M Circuit

- The external circuit resistance in both the E & M input and the E & M output must be such that the maximum DC current flow does not exceed 150mA under any conditions. Failure to observe this limit might mean that the protection devices on these leads do not unlatch after a transient event. Connecting the E & M input or E & M output to a power supply capable of supplying more than an amp may cause severe damage, due to overheating of EMC filter components in these circuits, should the protection devices trigger.
- Where the E & M output circuit is used to switch power to an inductive device, such as a relay, that device must have efficient suppression to absorb the inductive spike that occurs when the current is switched off. If the peak

spike voltage exceeds ± 58V the circuit protection devices might trigger and latch on. Where the relay is powered from 48V or more, only a silicon diode in parallel with the relay coil will provide a sufficiently tight clamping voltage. For lower switched voltages a Zener diode or MOV may be used provided its worst case clamping voltage does not exceed 58 V. RC transient suppression circuits should be used with caution: these require careful design to meet the suppression objectives.

## 4-Wire Audio Line Interface

- If DC is applied to the 4W audio lines, the nominal voltage should not exceed 48V. Under no circumstances should the peak voltage exceed 58V otherwise the protection devices may be triggered.
- As for the E&M leads, any DC applied to the 4-wire lines should be current limited, to no more than 150 mA, so as to allow the protection devices to unlatch after a transient event.
- The on-board protection devices are to be regarded as 'secondary protection' only. Generally they are only suitable for relatively benign environments such as internal building wiring or short run underground wiring. If more severe conditions are expected it is advisable to fit external primary protection devices eg. gas-discharge tubes or high-energy varistors between the external line connections and the network board line interface.
- Where there is a possibility of 'power-cross' conditions occurring, externally fitted fuses or PTC resistors are required to prevent any long duration high currents burning out components on the audio line interface. Such current limiting protection is mandatory for telco administrations requiring compliance to BellCore standard GR1089 or the power cross tests of UL60950 or ITU K.21.
- External Fuses or PTC resistors are also required if the equipment is required to be compliant with the current-limiting conditions specified for protecting external telco and customer premise wiring as per GR1089 and UL60950.
- Tait have used the Krone Comprotect 2/1-CP BOD190A1 product, part number 5909 1 078-40.

# 1 PPS Timing Reference Input

(Draft text. Not supported in firmware by base station version 1.13.)

- Where the cable length from the 1PPS distribution amplifier is short (< 3m) the internal 50 Ω termination in the network board is sufficient to suppress cable reflections. Multiple short cable feeds may be taken from a single distribution amplifier output provided that it has sufficiently low output impedance.
- Where the cable length exceeds 3m the distribution amplifier must have 50 Ω back-terminated outputs to suppress cable reflections. In such cases, only a single feed may be taken from each distribution amplifier output.
- (Unconfirmed) With properly terminated distribution amplifier outputs any length of cable may be used subject to the error introduced by the cable delay. It is recommended that no more than 20m of cable be used (equivalent to 100ns delay in cable with a 66% velocity factor cable).

- Failure to properly back-terminate longer cables at the distribution amplifier may result in the network board seeing jitter on the timing of the 1PPS pulse.

# Signaling on the Analog Line

Controlling a TB9100 P25 base station using an analog line connected console system presents some choices. The E & M interface may be used for TX key and RX call features, although these are readily carried by tone signalling.

## Tone Remote Signaling

- Low Level Guard Tone provides a means to key the transmitter when the dispatcher is talking.
- Function tones (Single or Dual) provide a means for dispatchers to control a base station - for example, to select a channel.
- Function tones may also select up to 16 channel profiles where P25 source and destination Ids may be set.
- Function tones may also be used as an input to task manager for very flexible base station control.

## MDC1200 Signaling

Tone remote can be combined with MDC 1200 to allow dispatchers to signal to a large number of selected terminals (analog and P25).

When sending from the console system to P25 terminals using MDC1200 PTTID, delays at the start of the over will occur because all of the call information needs to be available before the APCO Base Station is able to begin the P25 transmission. The P25 system passes the source and destination information mixed in with audio data.

Similarly when P25 terminal information is to be sent via the TB9100 to the console system using MDC1200 to identify the source and destination Ids, there is a delay issue with inband signaling. The TB9100 may be configured to send MDC1200 data at the start of the over, which may cause the beginning of the syllables of the call to be lost. Instead the TB9100 may be configured to send MDC1200 data at the end of the over, in which case the operators will only have calling ID available after the over has finished.

The TB9100 dispatch interface may use MDC1200 to map to these P25 features

| **MDC1200** | **Direction** | **P25** |
|---|---|---|
| Voice Alert | --> | Individual Call |
| PTT ID | --> | Group Call |
| Supplementary Services | --> | Supplementary Service, src,tgt (src = fixed). |
| ANI | <-- | Individual Call |
| ANI | <-- | Group Call |

For more details, see the Analog Line section of the Configuration part of the CSS manual or online Help.

# Syslog Collector

TB9100 base stations are able to send messages from their system log to any IP address on the network. This makes it easy to equip the TaitNet P25 network with a central syslog collector; any network-connected computer with third party syslog collector software will suffice. Other elements in the network such as routers and switches can also be configured to send syslog messages to the syslog collector. The CSS can also view the system log of the connected base station and save it to a file.

Computers running Unix or Linux have a syslog collector as part of their operating system. Windows-based PCs need a suitable third party syslog collector. Tait recommends the Kiwi Syslog Daemon (see [www.kiwisyslog.com)](http://www.kiwisyslog.com). It is able to handle the syslog messages of Cisco routers. The shareware version can be used to explore its capabilities, but the registered version offers useful additional functions such as the ability to display different screens for different base stations.

- Use the CSS to enable and configure the sending of messages to the Syslog collector (Configure > Alarms > Logging).
- Use a script to enable Kiwi Syslog to monitor base station failure. The script resets a timer if Kiwi Syslog receives a message with the base station's IP address. If the timer expires, the script carries out the action (emails the technician, pages the technician, or sends a syslog message).
- Kiwi Syslog can email the duty technician when significant error messages are received from a TB9100 or a router.

Syslog messages from the base station use the facility code `local0` (for base station messages) or `local1` (for call records). Tait suggests that routers use the facility code `local2` and that switches use `local3`. The core router can optionally be given its own facility code `local4`. This enables the syslog collector to efficiently separate messages from different sources.

For information about the syslog format used by Cisco routers, see Cisco's System Error Messages Overview:

[http://www.cisco.com/univercd/cc/td/doc/product/software/ios120/12supdoc/12sems/emover.htm](http://www.cisco.com/univercd/cc/td/doc/product/software/ios120/12supdoc/12sems/emover.htm) .

# Configuring Cisco Devices to Send Syslog Messages

For instructions on how to configure Cisco routers and switches to send syslog messages, see Cisco - Resource Manager Essentials and Syslog Analysis: How-To:

[http://www.cisco.com/warp/public/477/RME/rme_syslog.html#topic2](http://www.cisco.com/warp/public/477/RME/rme_syslog.html#topic2) .

In addition:

- Set the logging trap level to warnings rather than informational. (This is a higher level than the one Tait recommends for base stations, but it is not known how much information that level would add to router traffic. The number of base station messages will be small.)

- Add the following command to the configuration:

  For routers:

  ```
  logging facility local2
  ```

  Optionally, for the core router:

  ```
  logging facility local4
  ```

  For switches:

  ```
  logging facility local3
  ```

# Interpreting Tait Syslog Messages

Syslog messages have the format <PRI> HEADER MSG. (See the online Help for the Kiwi Syslog Daemon for more details and references.)

## PRI

PRI is the priority value. It consists of a facility code and a severity level. The default facility code used by TB9100s is Local0. All TB9100 alarms have a severity level of Warning or higher. The TB9100 generates a large number of syslog messages at the level of Notice. Only log to the syslog collector at this level if you are troubleshooting a problem that is otherwise difficult to track. Syslog messages are lost if the base station is reset. Also, when the base station's syslog message store is full, new messages overwrite the oldest.

Syslog messages logged to an IP address have a PRI but system log messages that the CSS gets from a TB9100 and stores to a file do not: they begin with the header.

## HEADER

The header in a syslog message consists of a timestamp and the hostname of the message originator. The TB9100 generates timestamps in the form:

YYYY-MM-DD'T'HH:mm:ssTZ

For example: 2004-06-24T12:05:43Z

This is the format recommended by RFC3339. The "Z" indicates that the timezone is UTC.

The hostname is in standard dotted quad format.

## MSG

The message in a syslog message consists of a tag, a colon separator, and the content.

### Tag

In TB9100 syslog messages, the tag has the format Module Code_Severity level_Mnemonic, for example:

ASF_4_SKSRV

The module code indicates which TB9100 module generated the syslog message. The severity level is the same as was in the priority level. The Mnemonic often indicates the process within the module that generated the message.

The following are the TB9100 module codes:

| **Module** | **ModuleCode** |
|---|---|
| TB9100 Base Station (General) | ABS |
| Analog Gateway (General) | AGW |
| Network board | ASF |
| Reciter | REC |
| Power Amplifier | PA |
| Power Management Unit | PMU |
| APCO Network Controller (not yet available) | ANC |

### Content

The content part of the message has the following structure:

Timestamp, Code – Text

For example:

00073877.134, 44 - Alarm NO_PA_DETECTED - STATE=ACTIVE

And:

69.074, BS clock changing NEWTIME=2005-03-01 03:41:07Z TIMESOURCE=172.25.111.91

The timestamp gives the value of the module timer since its last reset. It consists of a number of seconds and milliseconds. The code is not always present but uniquely identifies that message type. The text is an English description of the reason for the message. It may include variable values, written so as to be machine-parsable.

# Monitoring Base Station Failure

Every 15 minutes, TB9100 base stations send a "heartbeat" syslog message to the syslog collector, provided they are configured to log to the network at the Notice level and above. The MSG part of this message is like this:

1109798410.872, ABS_6_HRTBT: 00075816.412, 0 - Base Station Alarm is INACTIVE

You can set a syslog collector up to respond to base station failure as follows:

1. For each base station, set up a filter for any message from the base station's IP address.
2. Set up an action for that filter: if the syslog collector receives a message, it starts a timer.
3. Set up a duration for the timer of at least 20 minutes.

4. Set up an action if the timer expires (for example, send an email to the duty technician)

(If you are using the Kiwi Syslog Daemon, these capabilities are only available in the licensed version.)

# Numbering and Other Considerations

## NAC

Normally, base stations on a TaitNet P25 network are given the default NAC 0x293. The NAC is not used for selective squelch (i.e. to identify a talkgroup). There are the following exceptions:

- If there is more than one base station nearby that is on the same frequency, you will need to select another NAC.
- If users on other P25 systems that use NACs to identify talkgroups will be expected to interoperate with users on the TaitNet P25 network, it may be necessary to give the base stations the special receive NAC 0xF7F (accept any NAC).

## Talkgroup and Individual IDs

The talkgroup number range is 1-65535 (0x1 - 0xFFFF). The full individual number range is 1-167 777 215 (0x1 - 0xFFFFFF). Of these, 1-9 999 999 are available for general purposes, that is, suitable as radio IDs.

If the analog console system uses MDC1200 signaling, there are constraints on the available numbers. Tait recommends using the number ranges 1-4095 for groups and 1-61439 for individuals. These number ranges make the mapping between MDC1200 and digital P25 numbering schemes straightforward and correspond to the base station's default mapping. This mapping reflects the fact that console systems divide up the MDC1200 numbering range (1-65535: 0x1- 0xFFFF) between individuals and groups. Numbers beginning with 0xE are group numbers. If they were to be used as individual numbers, the console system might not permit individual operations such as radio disable.

If the analog console system uses MDC1200 signaling for supplementary services, there are additional constraints. The console system may not permit the use of MDC1200 numbers that have an E or an F in the hexadecimal version. The P25 equivalent of such numbers should not be programmed into radios.

## Preamble

While Tait P25 equipment has a fixed preamble, other subscriber unit radios may have a configurable preamble. Tait recommends giving them a preamble of around 20 ms. The preamble helps prevent late entry to voice calls and makes supplementary services (for example, call alert and status request) more reliable.