TaitNet P25 networks

System Manual

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Preface

Scope of Manual

This manual describes the TaitNet P25 network. It provides an overview of the network, describes the channel group and how it operates, discusses aspects of the network design, and gives practical assistance on how to manage an existing network.

The manual is primarily concerned with conventional networks. However, channel groups or individual TB9100 base stations can be part of a P25 trunked network with a third-party trunking site controller. The manual describes the operation and configuration of channel groups within this kind of trunked network.

Document Conventions

"File > Open" means "click File on the menu bar, then click Open on the list of commands that pops up". "Monitor > Module Details > Channel Module" means "click the Monitor icon on the toolbar, then in the navigation pane find the Module Details group, and select Channel Module from it".

Within this manual, four types of alerts are given to the reader: Warning, Caution, Important and Note. The following paragraphs illustrate each type of alert and its associated symbol.

Warning!! **This alert is used when there is a potential risk of death or serious injury.**

Caution **This alert is used when there is a risk of minor or moderate injury to people.**

Important This alert is used to warn about the risk of equipment damage or malfunction.

Note This alert is used to highlight information that is required to ensure procedures are performed correctly.

Associated Documentation

The full customer documentation set for TaitNet P25 networks is provided on the product CD supplied with TB9100 base stations. Updates may also be published on the Tait support website. Technical notes are published from time to time to describe applications for Tait products, to provide technical details not included in manuals, and to offer solutions for any problems that arise.

Publication Record

Issue	Publication Date	Description
1	February 2007	First release

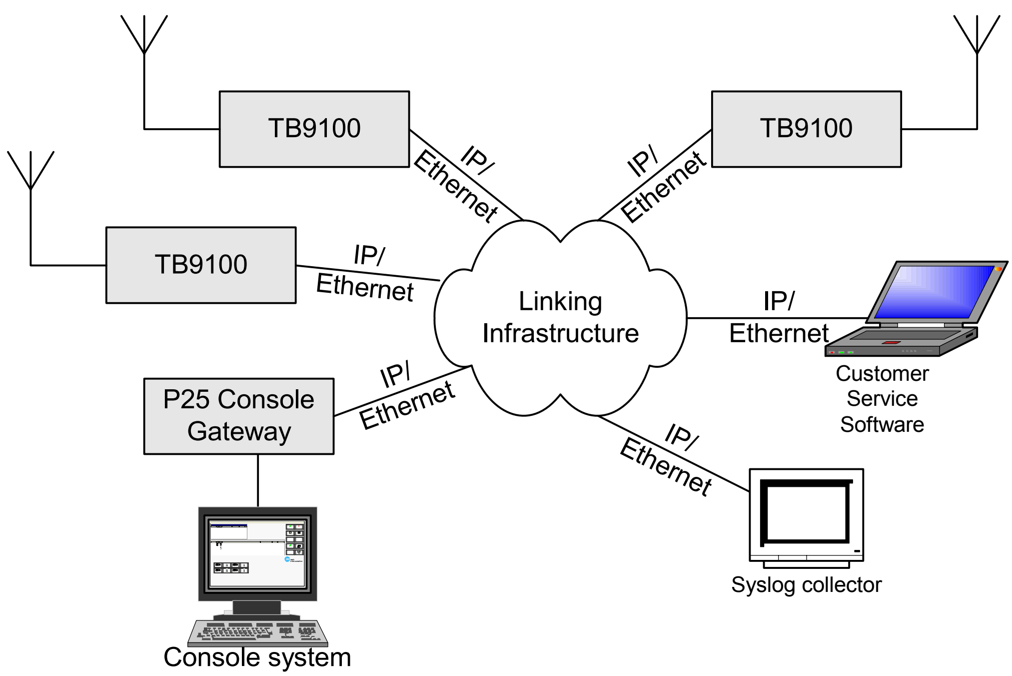
1 Network Overview

A TaitNet P25 network is a set of interconnected TB9100 base station transceivers. Each TB9100 can receive from and transmit to mobile and portable radios, just like any base station/repeater. However, TB9100s also have built-in networking capabilities that enable them to combine together to form one or more logical channels with wide area coverage.

When a P25-capable 2-way radio (referred to as a subscriber unit or SU) makes a call, a TB9100 receives it and passes it over the linking infrastructure to other TB9100s, which can repeat the transmission.

Third-party dispatch console systems can be integrated with TaitNet P25 networks. Often, they connect via a Tait P25 Console Gateway. P25 Console Gateways have the same networking capabilities as TB9100 base stations, but they have no RF capability. They can serve as an encryption/ decryption point to enable analog dispatch consoles to participate in encrypted calls.

TB9100 base stations and P25 Console Gateways are the main network elements in a TaitNet P25 network. They have voting capabilities and an Ethernet interface, so that extra modules such as voting comparators and digital interfacing equipment are not required. They are interconnected over an IP-based linking infrastructure.



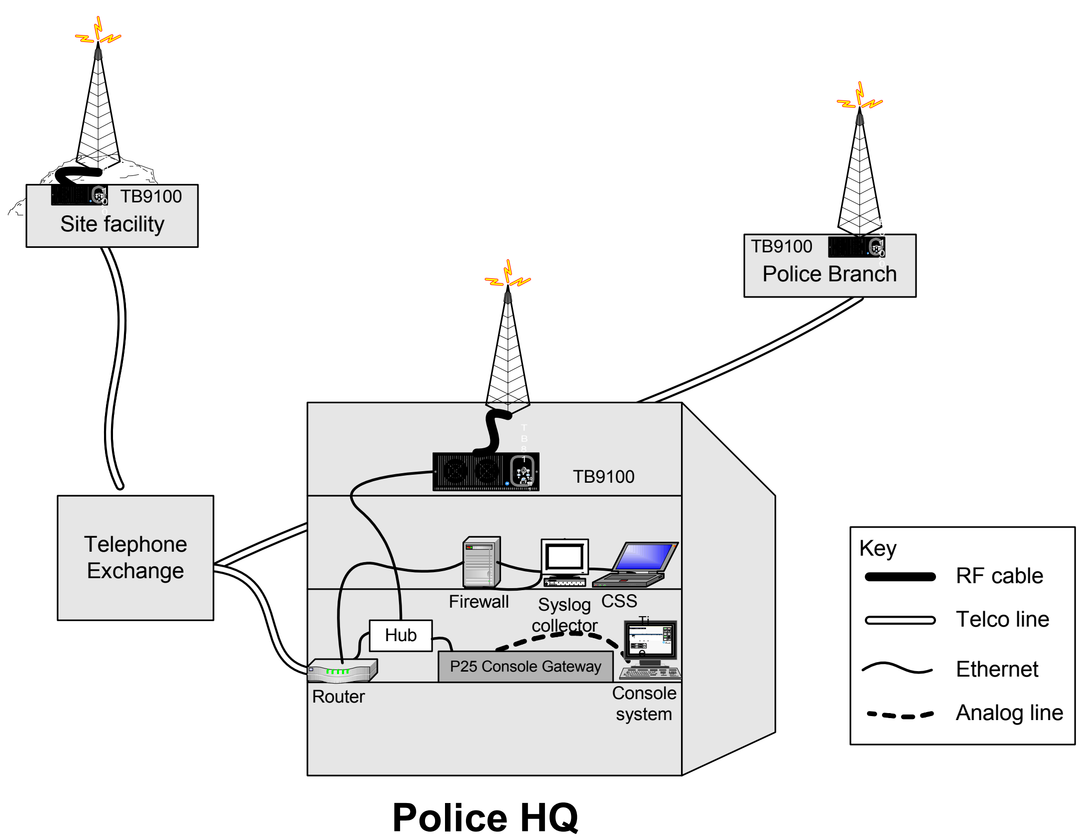
The TaitNet P25 network is managed using the Customer Service Software (CSS) and a syslog collector. The CSS can connect to any Tait network element from anywhere in the network or beyond. It can remotely monitor and configure the connected network element. It can also carry out diagnostic tests and upgrade firmware. A third-party Syslog collector acts an an alarm center, receiving alarms and call records from any network element, displaying them, and storing them in logs.

The linking infrastructure passes voice over IP and signaling messages between network elements. It also carries CSS and syslog communications.

TaitNet P25 networks comply with the APCO P25 set of standards. This means, for example, that they support dual mode (digital P25 and analog FM) operation and the use of P25-compliant mobiles and portables from other manufacturers.

1.1 Example Installation

In the following simple example of an installed TaitNet P25 network, analog dispatch equipment is connected via the base station's analog line. Two channels are located remotely at different sites, resulting in a star topology. This means that there are never more than two hops from one base station to another. As all three channels operate as one, conversations are automatically transmitted from all the base stations. The TaitNet P25 network is linked via a firewall to the police LAN. The maintenance technician's PC running CSS and another PC serving as a syslog collector are part of that LAN.



1.2 Linking Infrastructure

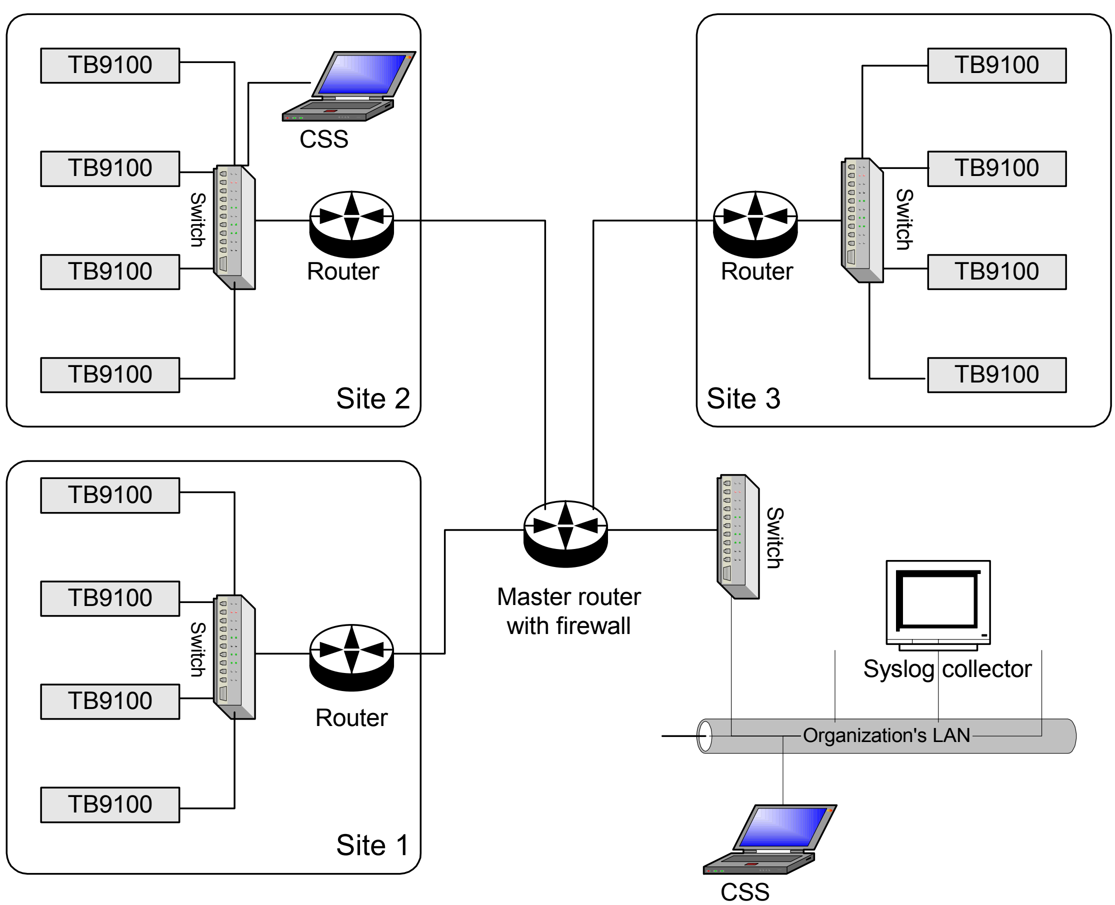
The linking infrastructure is what interconnects the TB9100 base stations to form a TaitNet P25 network. It is IP-based and consists of LAN equipment at each site and a bearer network of links that interconnect the sites. The LAN equipment belongs to the TaitNet P25 network, while the bearer network can be leased from a telephone company.

The linking infrastructure can be integrated with an organization's LAN. This makes it possible for PCs running CSS or a syslog collector to connect from anywhere in the organization via the existing LAN. However, access control or a firewall should be provided for security and to restrict the traffic on the bearer network to TaitNet voice, CSS, and (in the future) subscriber unit data.

There are two main types of linking infrastructure. Routed networks are WANs that connect sites using routers. They are suited to low-bandwidth links. Switched networks are LANs. They have a lower delay but require high-bandwidth links. Switched networks only use routers for linking the TaitNet P25 network with the organization's LAN.

Routed Network

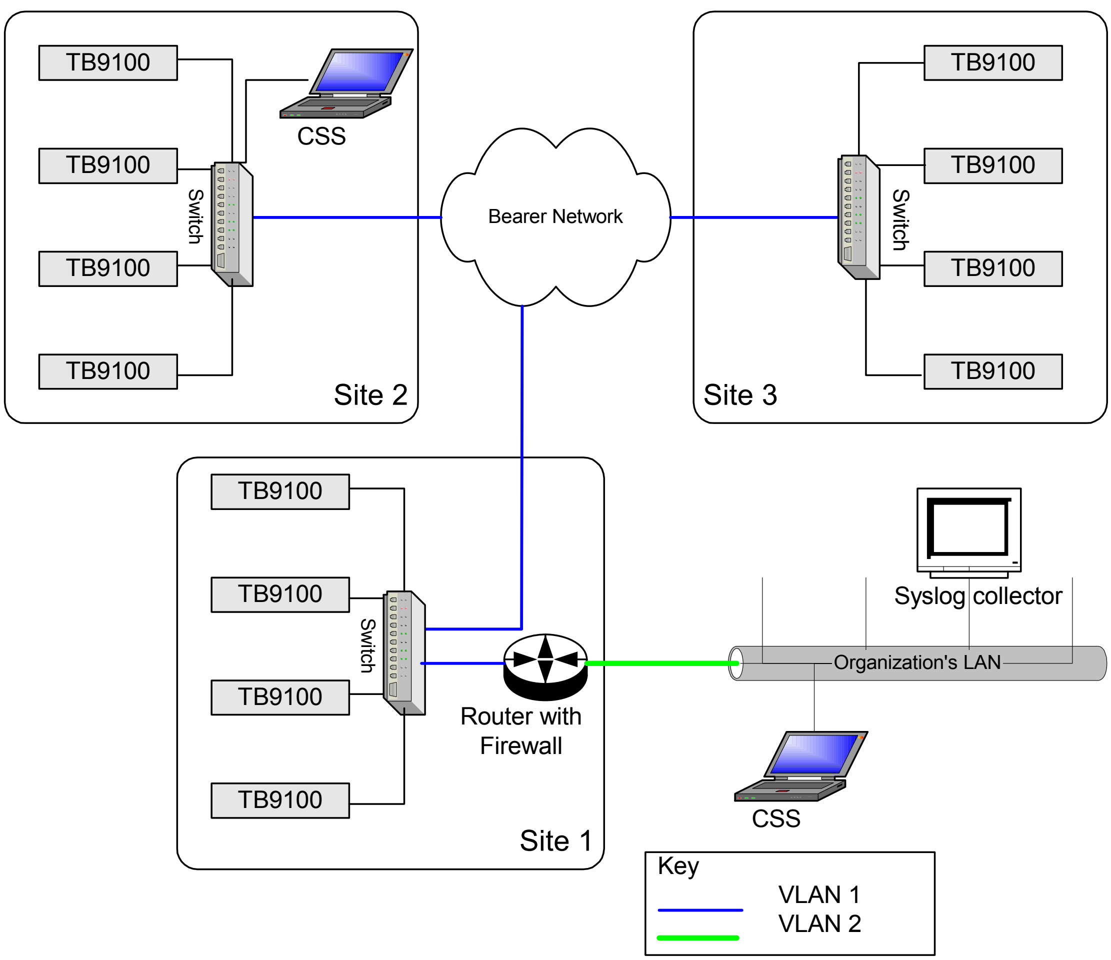
In routed networks, each site has its own LAN segment. Routers are the interface between the LAN segments and the bearer network. They convert the data between Ethernet and other protocols such as T1 (for feeding to the bearer network) and vice versa.



This example uses a star topology, which limits the number of router hops to two even with a larger number of sites. Only Tait-approved routers can be used and they must be configured for particular requirements such as IP multicast and Quality of Service (to give the voice stream priority over other data). The bandwidth requirements can be reduced by the use of compressed RTP.

Switched Networks

In switched networks, switches are used instead of routers to link sites. The whole TaitNet P25 network is essentially a single LAN. This is achievable if the remote links between sites have a sufficient bandwidth. The TaitNet P25 network can be integrated with the organization's LAN by configuring each as a separate VLAN (virtual LAN). Then a router is needed to route data between VLANs, in this case to enable the CSS and the syslog collector to operate from within the organization's LAN.



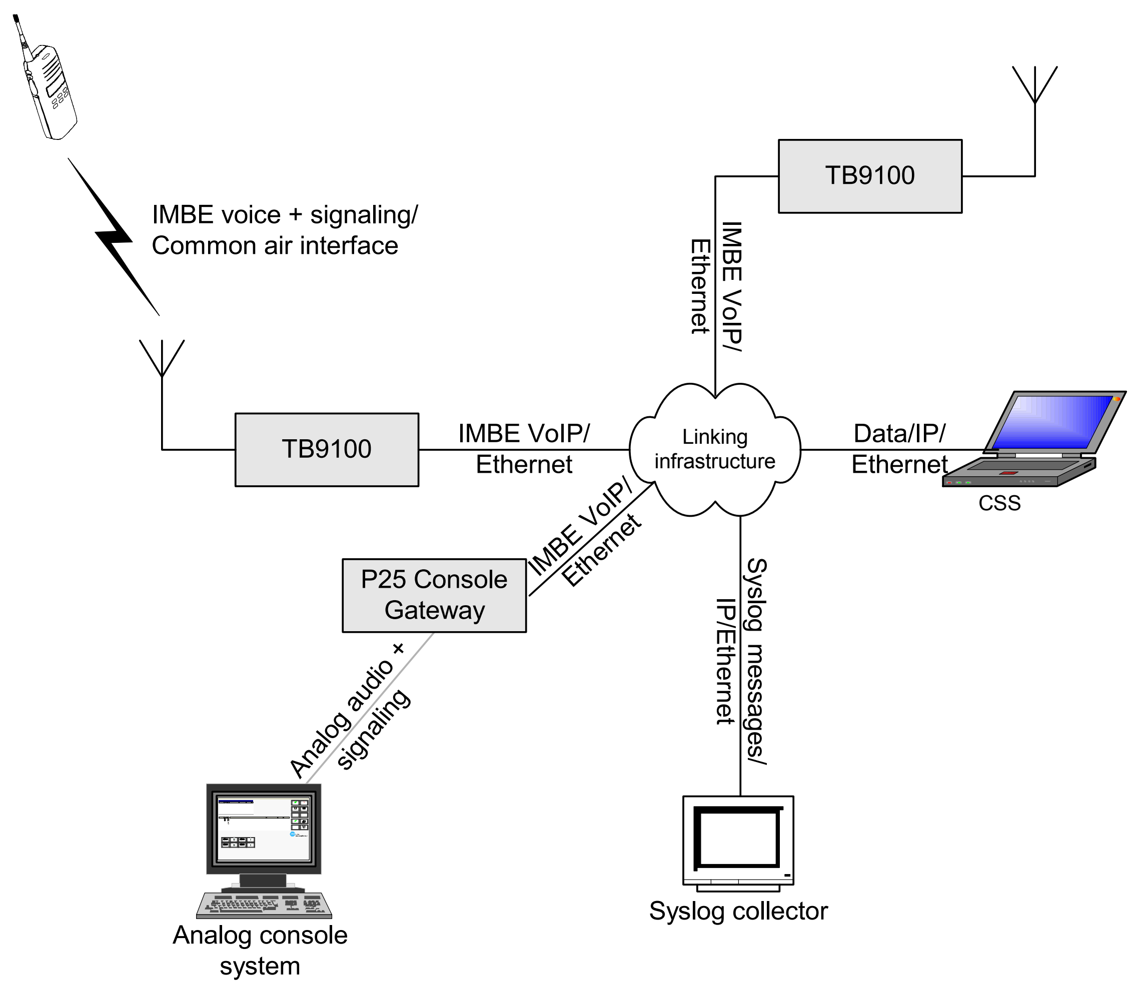
The links between switches need to have sufficient bandwidth to ensure minimal jitter. Switches are not normally able to prioritize voice over IP packets.

1.3 Network Signal Paths

The base stations receive calls from SUs. In digital P25 mode, these communications are in the IMBE format, with forward error correction added. The base stations correct any errors and remove the forward error correction bits. They put the result on the network as 'voice over IP' using RTP (the Internet real-time protocol). Routers or switches pass the RTP packets to the router, and then over the bearer network to other network elements.

P25 Console Gateways convert these communications to suitable protocols and pass them to the dispatch console equipment. If this equipment is digital, the DFSI protocol is used. If the equipment is analog, voice is converted to analog and any subscriber signaling can be converted into its MDC1200 equivalent.

Analog FM calls can also be networked. The received voice is put into the G.711 digital format and sent out using RTP. G.711 needs more bandwidth than IMBE.



In addition to this voice traffic, the linking infrastructure carries alarm messages and CSS communications. Alarms and other status indications can be sent to a syslog collector. A PC running the CSS can communicate with any base station on the network, monitoring it, changing its configuration, or carrying out diagnostic tests.

1.4 Dual Mode

The TaitNet P25 network supports dual mode operation. TB9100s can receive and transmit in digital P25 and in analog FM mode and the network can carry speech for both modes. If a TB9100 receiver's RF interface is configured for dual mode, it listens for digital P25 signals. If it detects them, it switches to digital P25 mode, otherwise it receives in analog FM mode. When a dispatcher initiates a call, the TB9100 transmits in the default mode for the channel, unless the console selects a different calling profile.

Base stations operating in dual mode operate to their full specification, even if the analog FM mode is wide-band. An additional filter in the digital front end means that the receiver can listen for analog FM with a wide-band setting and for digital P25 using a narrow-band filter. Sensitivity and selectivity for digital P25 signals are largely unaffected by dual-mode operation.

1.5 Channel Groups

TaitNet P25 networks consist of one or more channel groups. Each channel group is a single logical channel that provides a distributed RF receive and transmit function. Typically, a channel group contains one base station from each site in the TaitNet P25 network. This means that the channel group coverage extends over the entire network area. Channel groups can interface to digital dispatch equipment, to analog dispatch equipment, and to a trunking site controller. The dispatch equipment often interfaces via a P25 Console Gateway. A base station can only be a member of one channel group at a time.

Channel groups are implemented using IP multicast. To assign a base station to a channel group, you configure it with the channel group's multicast address. When an inbound RF signal arrives at a base station, it is sent to the multicast address. All channel group members are listening to that address; they receive the signal from the network. If their repeat function is enabled, they broadcast the signal.

If more than one signal arrives at once, a voting and switching process determines which signal the channel group transmits. This is carried out by the channel group itself. No external equipment is required.

1.6 Dispatch Console Support

Dispatchers communicate over a radio channel with SUs. TaitNet P25 networks facilitate this communication. Dispatch equipment can be line-connected to a single TB9100 base station or to a whole channel group. If there are multiple dispatcher positions, the dispatch equipment usually has some kind of console switch to enable the dispatcher to select the channel to talk on.

Often, dispatch equipment is connected to the TaitNet P25 network via a P25 Console Gateway. The P25 Console Gateway provides an analog line if the dispatch equipment is analog or a DFSI interface if the equipment is digital. Alternatively, dispatch equipment can be connected directly to a base station in the channel group. With the appropriate feature licenses, any TB9100 can provide an analog line interface or a DFSI interface.

Analog Console Systems

The TaitNet P25 network supports line signaling, tone remote function tones and Motorola MDC1200 signaling. During analog FM calls, the network is transparent to analog signaling; analog voice and signaling is converted into the G.711 format for transport across the IP network. During digital P25 calls, any analog signaling from the console is mapped to the P25 standards for transport across the air interface.

With the appropriate feature licenses, the P25 Console Gateway can function as an encryption/decryption point.

Digital Console Systems

Digital console systems connect to a channel or channel group using the DFSI interface defined in the P25 TIA standard. Encryption and decryption is carried out by the console system. The DFSI interface supports analog FM mode.

1.7 Encryption

All TaitNet P25 networks support end-to-end encryption between radios. No additional licenses or configuration settings are required. The network simply passes on the encrypted data or voice.

If analog dispatch equipment needs to participate in encrypted calls, a P25 Console Gateway is needed to provide an encryption/decryption point.

1.8 Network Management

Using the Tait CSS software, you can remotely connect to any Tait network element and monitor or configure it. The CSS can monitor the current status of all alarms in the connected network element and display its call record log or system log, containing alarms and other messages. The CSS can also monitor voting in a channel group, conduct diagnostic tests, and upgrade firmware.

The TaitNet P25 network can also be set up with a syslog collector. This can receive alarms and call records from any network element, displaying them and logging them to file. It can also monitor heartbeat messages. In this way, you can monitor the whole network from a single location.

1.9 Uplink Voting, Downlink Voting, and Simulcast

When a network does not have voting or simulcast, users must manually change channel as they move from the coverage area of one base station to another. This cannot be done without terminating the call and starting another.

In conventional networks, uplink voting, downlink voting, and simulcast are options that automate the handover from one base station to another.

With uplink voting, the base stations in a channel group are all on the same receive frequency. The channel group compares all the signals received and provides the best signal to the dispatcher and (if RF repeat is enabled) to the channel group transmitters.

With downlink voting, the SUs rapidly sample the signals from the channel group transmitters and select the best signal. The base station transmitters are configured to transmit a preamble before the call itself, and to begin transmitting at approximately the same time (otherwise SU voting would compare different signals and give misleading results). The SUs are configured to continuously scan their list of channels looking for activity. When activity is detected, the RSSI of all channels on the list is measured, and then the SU switches to receive the one with the best RSSI.

Uplink and downlink voting combine to completely automate the process of channel selection. Dispatchers and SU users can listen and speak without needing to first switch to the correct channel.

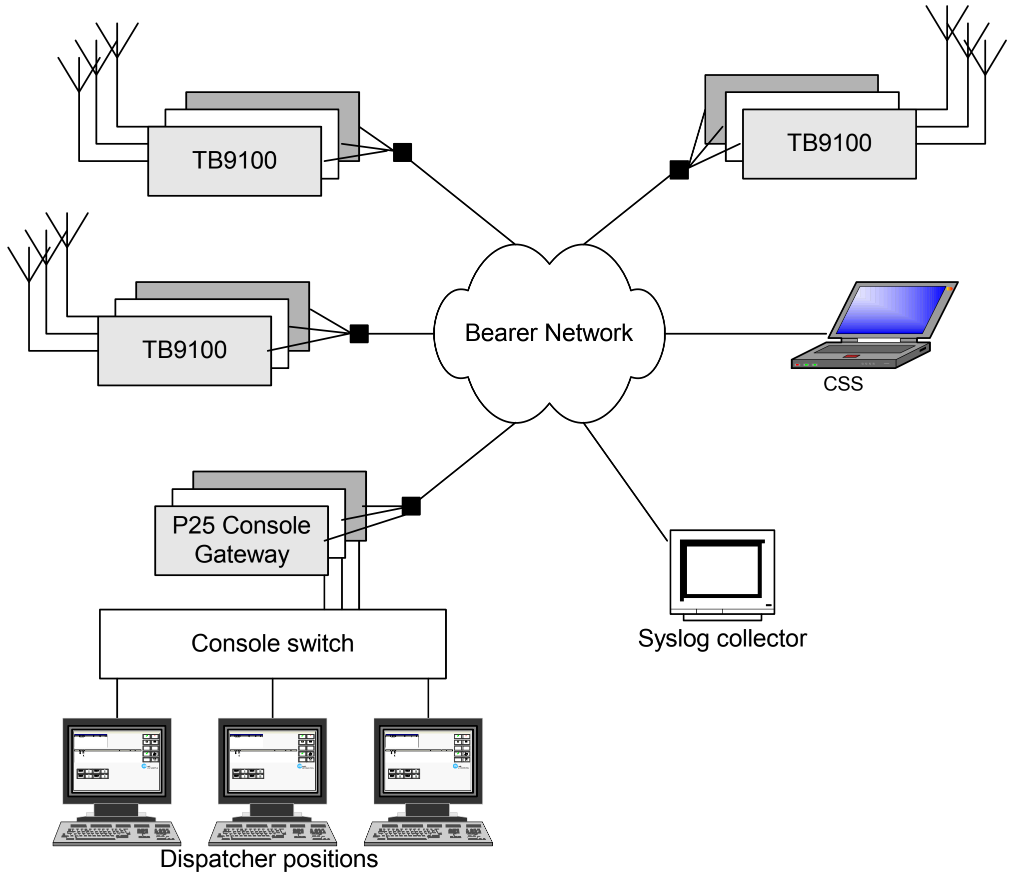
Downlink voting does add to the voice delay of the system. The more channels to be voted on, the greater the delay. Refinements in SU voting procedures can mitigate the increase in voice delay but not remove it. To eliminate manual channel switching without adding to the downlink voice delay, simulcast is required.

Simulcast (planned for the version 3.1 release of the TB9100 base station) is the simultaneous broadcast of the same signal from several base stations, all of which transmit on exactly the same frequency. Simulcast is combined with uplink voting, which must be centralized in one channel group member, so that all voice streams can be time-stamped at a single location. This makes possible the synchronizing of signal transmissions. With simulcast, SUs always receive on the same frequency. For most of the coverage area, one base station transmitter has a significantly greater signal strength and 'captures' the SU receiver. In overlap areas where transmitters have similar strength, there can be a degradation of signal quality but this is minimized by tight control of the carrier frequency and the modulation fidelity.

1.10 Standard Conventional TaitNet P25 Network

A conventional TaitNet P25 network often consists of several sites and many channels, allowing several channel groups to co-exist in the one network and supporting several dispatcher positions. In the example below, there are three channel groups, each having one TB9100 at each of three sites. As the linking infrastructure consists of third-party equipment that forms the building blocks of the Internet, it can easily be scaled up as needed.

Each channel group can be connected via a P25 Console Gateway to an analog console. Alternatively, any channel group member can have a DFSI interface to a digital dispatch console. The channel group design supports the connection of additional consoles, to provide redundancy.



If the base stations in a channel group have the repeat function enabled, that channel group acts as a wide-area repeater (however, dispatcher calls have priority over radio calls for transmitting over the air).

If the repeat function is disabled, the channel group acts as a wide-area line-connected base station. Calls received by any base station are sent to the dispatcher and calls from the dispatcher are transmitted via all the base stations.

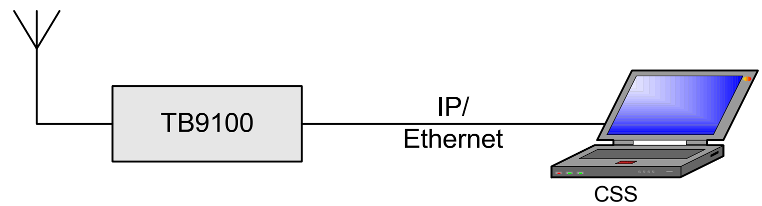
1.11 Other System Designs

TB9100 base stations and P25 Console Gateways can be part of systems other than a standard TaitNet P25 network. The following examples show some of the options, beginning with the simplest.

Simple Repeater

A single TB9100 can be configured to operate as a simple P25-compliant repeater. Its Ethernet interface is only used for communications with the

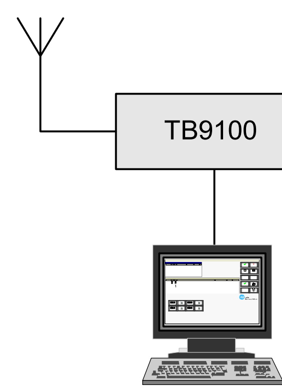
CSS. The CSS can be connected using either an ordinary or a cross-over Ethernet cable.



If remote access is required, the installation needs a router and a switch.

Single Base Station

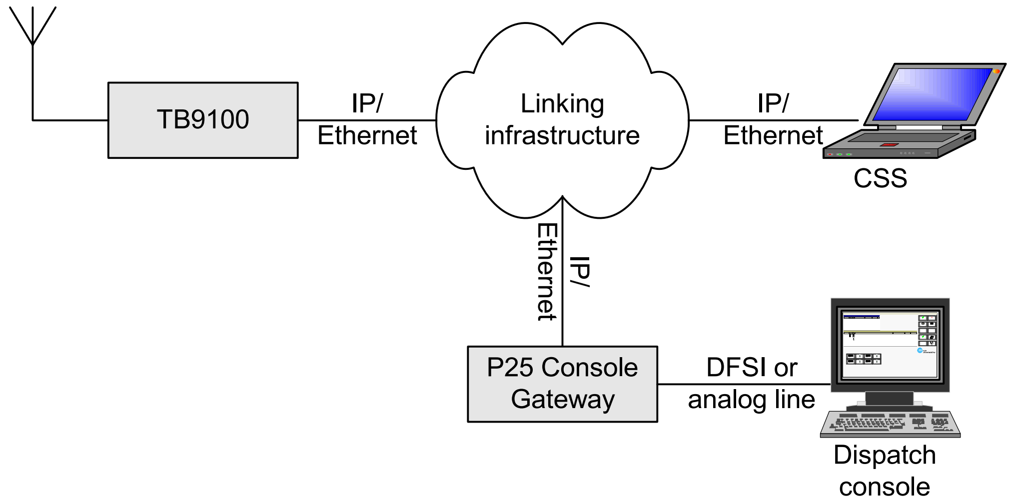
A TB9100 can be configured to operate as a simple P25-compliant base station, connected to a third-party dispatch console. If the console is analog, it connects to the TB9100 via the analog line. If the console is digital, it connects via DFSI over IP.



With an antenna relay, the base station can operate in simplex mode. This only requires a single antenna and one operating frequency for both transmitting and receiving. The repeat function must be disabled.

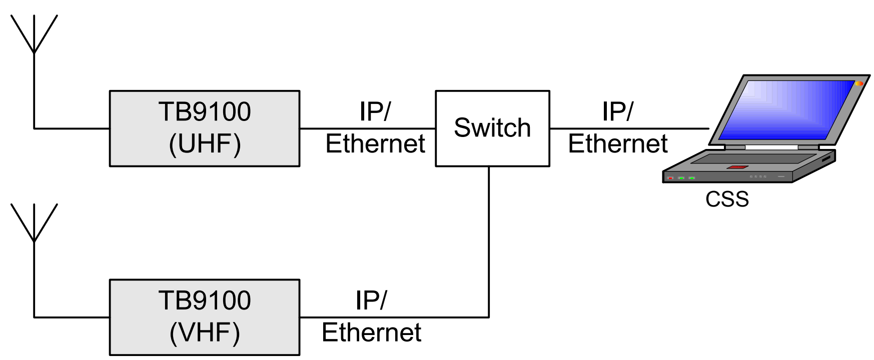
Single Base Station with P25 Console Gateway

Third-party console equipment can be connected to the base station via a P25 Console Gateway. If the console equipment is connected to the gateway via an analog line (analog equipment) or via the DFSI (digital equipment). The P25 Console Gateway communicates with the TB9100 using proprietary Tait channel group communications (the TB9100 and the P25 Console Gateway are effectively a channel group).



Cross-Band Repeater

The TB9100's built-in channel group interface makes creating a cross-band repeater straightforward. Two base stations (for example one UHF and the other VHF) are joined via a switch and are configured to belong to the same channel group. They can operate in both analog FM and digital P25 modes, but do not provide a cross-mode capability (analog FM to digital P25 or vice versa). This topology can easily be extended to provide cross-band repeating between three frequency bands.



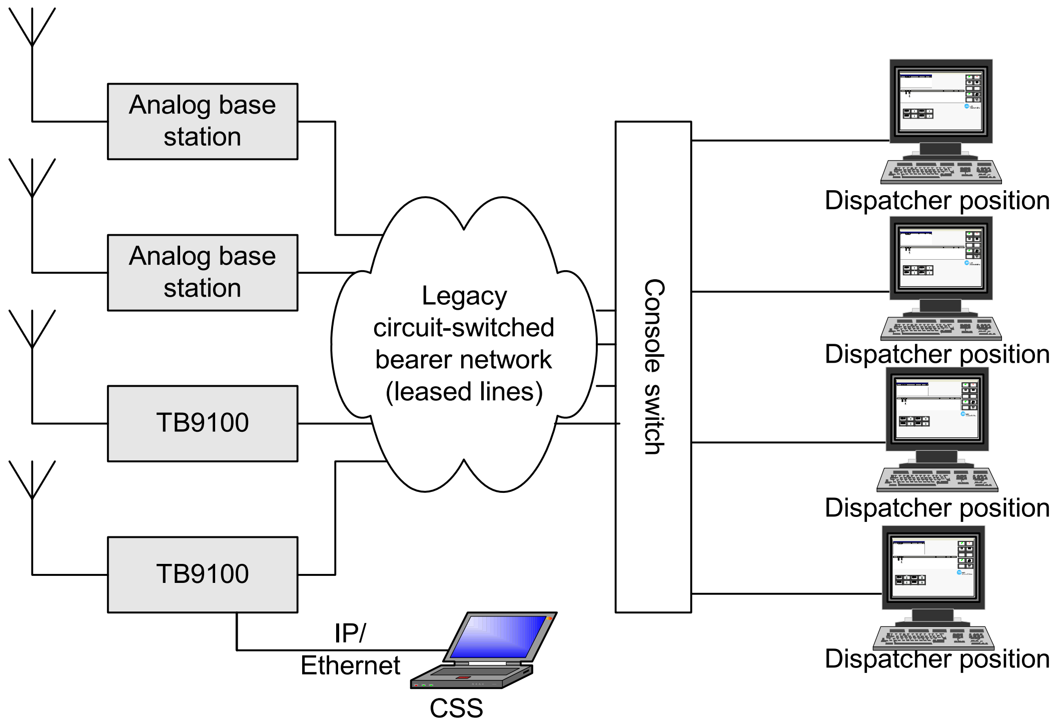
Cross-Mode Repeater

In a cross-mode repeater, one TB9100 operates in digital P25 mode and another TB9100 (or legacy base station) operates in analog FM mode. The base stations must be connected by analog line. This is because channel groups do not support converting a call between analog FM and digital P25 modes. E & M signaling is required to key the transmitters.

TB9100 base stations can be connected by their analog lines for other purposes as well. This may be cost-effective if there is no existing IP-based infrastructure, but voice quality is reduced and signaling information is lost.

Legacy System with TB9100 Base Stations Added

TB9100 base stations can be added to existing systems. The existing bearer network can be used. The bearer circuits connect the console system to the TB9100s via their analog interfaces. In this topology, there is no IP-based network, although one could be set up to provide remote CSS access to the TB9100s. The TB9100s can operate in analog FM or digital P25 mode.



Trunked Wide-Area Channel

A channel group can function as a wide-area traffic channel or control channel within a trunked network. Instead of connecting to a channel consisting of a single base station, the trunking site controller connects to a whole channel group, giving the channel wide-area coverage.
(For availability, contact Tait.)

1.12 Making the Transition to Digital

The dual-mode capability of Tait TB9100 base stations gives network providers flexibility when making the transition from an existing analog conventional system to a new digital P25 network. TB9100s can replace analog base stations and interoperate with analog FM radios. Here are some possibilities:

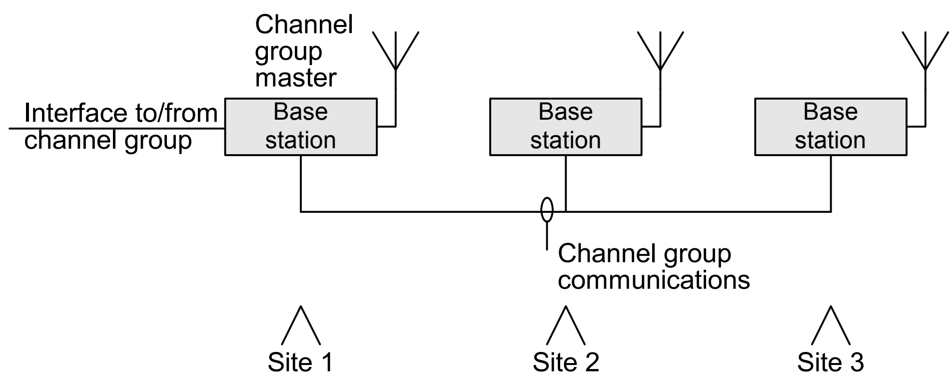
- Add P25 radios and/or TB9100 base stations to the existing system. Operate initially in analog FM mode, but change later to digital P25 mode.

- Add digital P25 channels to the existing analog system. This lets the network provider begin with P25 while continuing the existing conventional capability.
- Set up a TaitNet P25 network, but configure at least some of the TB9100s to operate in analog FM mode, for the support of legacy analog radios. Purchase P25 radios as new or replacement radios are needed. When a talk group has all P25 radios, it switches to digital P25 mode for normal operation but can always change channel to interoperate with analog-only radios. When all groups use digital P25, change the base stations to operate only in digital P25.

For more details, see “Making the Transition to Digital” on page 67.

2 Channel Group

TB9100 base stations are designed to work together as channel groups. A channel group is a single logical channel that provides a distributed RF receive and transmit function. Often, a channel group has one member at each site in the network, so that it can provide coverage over the full network coverage area.



The members of a channel group communicate with each other using an IP-based proprietary Tait protocol. If desired, the channel group can have a voting function enabled by software. This voting provides the best available RF signal for repeating and providing to external interfaces.

A channel group member can provide a gateway function, offering an analog line interface, a trunking (TCCP) or a digital dispatcher (DFSI) interface to an external device. This member is referred to as the channel group master, as it exerts control over the channel group, based on commands it receives from the external device.

P25 Console Gateways can be members of the channel group just like TB9100 base stations. A channel group can have up to 14 members.

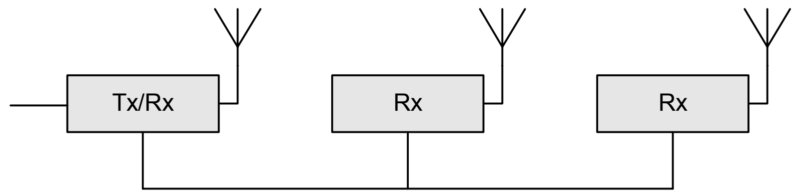
2.1 Types of Channel Group

Channel groups of different types can be used in conventional and in trunked networks.

Single Transmitter with Fill-in Receivers

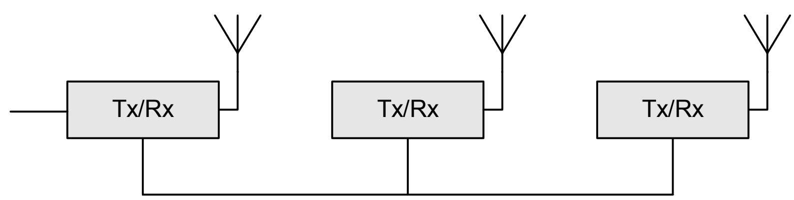
A channel group makes it easy to extend the coverage of a base station by adding a number of strategically placed fill-in receivers. The transmitting base station is usually but not necessarily the channel group master.

Continuous voting selects the best available RF voice stream for repeating and for providing to the dispatcher.



Multiple Transmitters — with SU Voting or Scanning

The channel group can have multiple transmitters. In conventional systems, SUs can be configured to scan the frequencies of the channel group transmitters and select the best downlink signal. Channel group receiver voting continually selects the best uplink signal for providing to the dispatcher and to other base stations for repeating.



Multiple Transmitters — with Simulcast

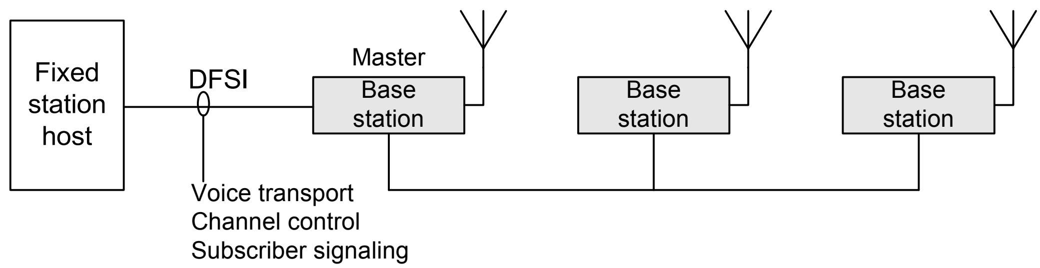
In a simulcast network, the channel group transmitters all use the same downlink frequency. This is a useful option where channel frequencies are scarce or expensive. SUs do not need to scan, reducing the call setup time. Simulcast makes sure that the transmissions are on exactly the same frequency and precisely synchronized. The base stations are also configured to use the same receive frequency. Continuous channel group voting selects the best uplink signal for providing to the dispatcher and to other base stations for repeating.

2.2 Channel Group Applications

Digital or analog dispatch equipment can be connected to a channel group. From the dispatch equipment's perspective, the channel group is just like a single channel, except for the wide area coverage. Analog dispatch equipment connects via a P25 Console Gateway. A channel group can also function as a traffic channel or a control channel in a trunked network.

Channel Group with Digital Dispatcher

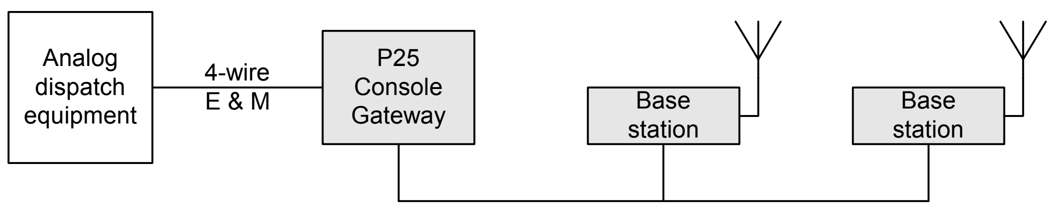
Digital dispatch equipment is connected to the channel group master. This can be a base station or a P25 Console Gateway. The master interfaces between the digital dispatch equipment and the channel group. Communication is via the P25 digital fixed station interface (DFSI). The dispatch equipment is known as the fixed station host (FSH). All communications between the channel group and the FSH pass through the master. The master and the FSH communicate using the DFSI protocol defined by the P25 standard. The master also communicates with the other members of the channel group. (IP allows the use of distinct protocols on the same physical interface.)



The DFSI provides voice transport, channel control, and subscriber signaling services.

Channel Group with Analog Dispatcher

Analog dispatch equipment can also be connected to a channel group. The attachment is via a P25 Console Gateway or via the analog line of a base station. This converts between analog and digital P25 voice and signaling. The P25 Console Gateway can also serve as the encryption/decryption point for dispatcher communications.



The connection between the analog dispatch equipment and the P25 Console Gateway is the analog equivalent of the DFSI.

- Subscriber signaling (optional) is MDC1200 (also known as Stat-Alert™)
- Voice transport is 600 Ω 4-wire
- Channel control is tone remote and/or E & M line signaling

Channel Group with Trunking Site Controller

Channel groups can also be used in trunking systems. One member acts as the master, passing communications between the third-party trunking site controller and the channel group using the proprietary TCCP protocol. Under site controller instruction, the channel group functions as a wide-area control channel or traffic channel.

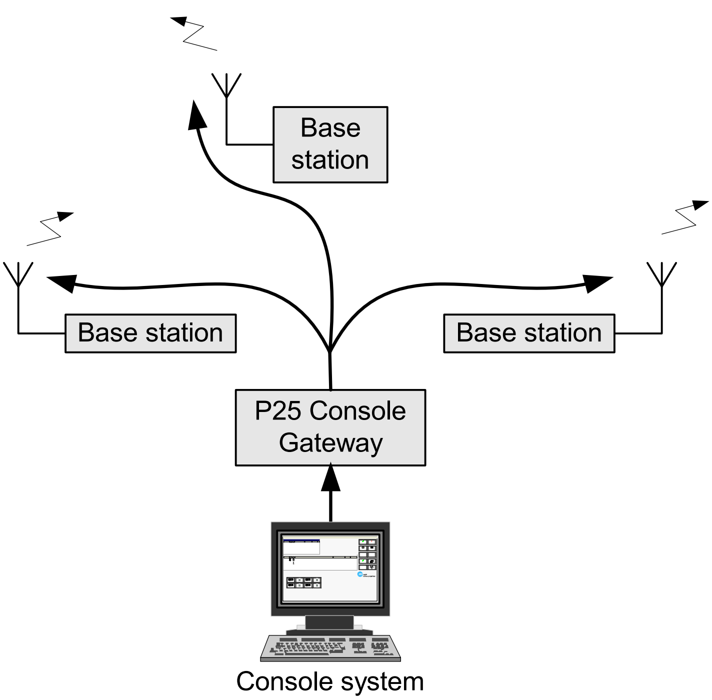
2.3 Channel Group Operation

How do the members of a channel group work together as a single logical channel? When a signal appears at an interface, the member turns it into a voice stream.

Voice streams are turned into RTP packets for sending to other members of the channel group. They are sent to a special type of IP address, a multicast address. All members of the channel group listen to the same multicast address. All of them are able to receive the RTP packets on their digital line and to transmit the digital voice signals contained in them.

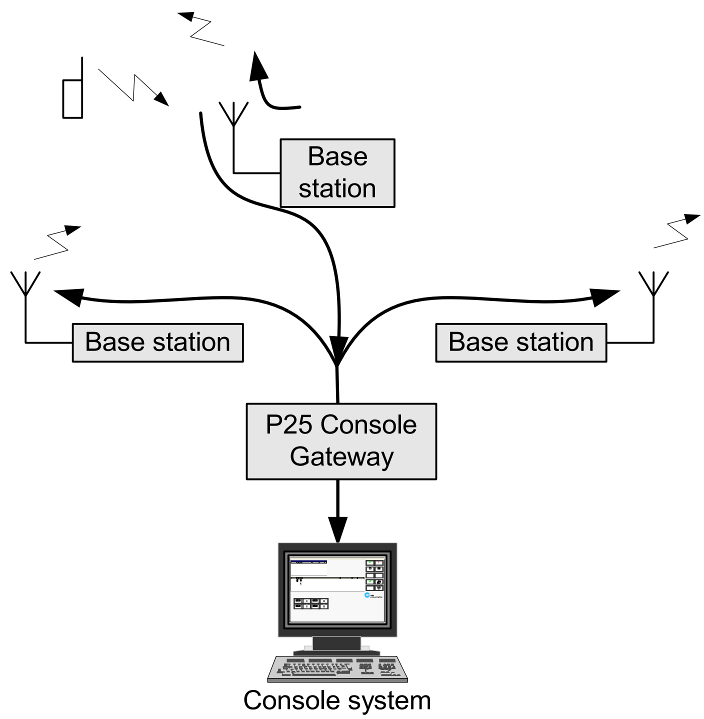
Dispatcher call handling

For example, when a dispatcher initiates a call, the P25 Console Gateway turns the call into a voice stream and puts it as RTP packets on the TaitNet P25 network. The other members of the channel group receive the packets and transmit the call over the air.



SU call handling

Similarly, if a SU transmits, a base station receives the signal, turns it into a voice stream and sends it as RTP packets to the channel group's multicast address. The P25 Console Gateway receives those packets and turns them into analog voice, which it sends to the analog dispatch equipment. If the base stations in the channel group have RF repeat enabled, they re-transmit the call.



Handling call contention

If multiple signals come from different interfaces or different callers at the same time, a selection process chooses the voice stream with the highest priority (see "Selecting a Voice Stream" below). If multiple signals appear at the RF interface from the one caller, voting continually selects the voice stream with the best quality (see "Voting" on page 28). Each channel group member has a switch for making selecting and voting decisions.

Selecting a Voice Stream

If the channel group receives more than one call at once, a selection process chooses one of the calls. These calls can arrive at different interfaces. Each channel group member selects the voice stream with the highest priority.

The selection process is quite complex. Voice streams are classified into different types, depending on whether they originate from a dispatcher (usually an analog line or DFSI), a SU user (usually an RF interface), or the maintainer (usually a control panel microphone). Each outgoing interface applies its own rules. This means that dispatchers will get to hear a calling SU in preference to another dispatcher, but SU users will hear the dispatcher in preference to another SU user. Calls higher on the priority table can also pre-empt existing calls. For example, if a call from a SU user is being repeated, and a dispatcher call is started, the channel group will transmit the dispatcher call.

Selection criteria

If voice streams have the same priority, P25 calls are selected in preference to analog FM calls. If the calls are the same type, the voice stream from the channel group member with the lowest receiver number is selected. If there are no receiver numbers, IP addresses are used instead.

Voting

The channel group itself is able to continually vote at packet level on the voice streams it receives from its RF interfaces. This enables it to repeat the best possible received signal and to provide that signal to the channel group's interfaces. In effect, voting blends or combines multiple voice streams, always picking the best packet out of those on offer. Voting is carried out by the same switches that handle the selection process described above.

Voting can be centralized at one channel group member or distributed among all members. If the channel group receivers use different frequencies, there is no voting; the receivers are configured with the voting type 'switched.' Using the DFSI, digital dispatch equipment can override the channel group's normal voting operation. It can select the voice stream from any channel group member or disable any member so that it does not participate in voting.

Centralized voting

In centralized voting, one channel group member carries out the voting. All voice streams are sent to that member. P25 voice streams originating at an RF interface are voted on every voice codeword (20 ms), resulting in a high quality voice stream. Analog FM voice streams are voted 5 times per second. Voice streams that didn't originate at an RF interface also pass through the central voter's switch.

A skew (differences in the arrival time of different streams from the same caller) of up to 60 ms can be compensated for in digital P25 mode but not in analog FM.

Any member of the channel group can be the central voter. In a channel group with fill-in receivers, making the transmitting base station the central voter shortens the average signal path.

To avoid having a single point of failure, a second channel group member can be configured for central voting. It will take over as the central voter if the first member fails. When more than one member is configured as a central voter, the member with the lower receiver number becomes the central voter. The other member operates as a satellite voter.

Centralized voting is required for simulcast networks. It is the best option for star topologies. It gives optimal results if the longest network delay from a member to the central voter is <40 ms and reasonable results if the longest delay is <80 ms.

Distributed voting

When voting is distributed, all the base stations in the channel group that are receiving valid RF signals vote in concert. If a base station wins the vote, it sends its voice steam to the channel group. If it loses the vote, it ceases sending its voice stream to the channel group, to save linking bandwidth. Voting occurs every LDU (180 ms).

Distributed voting may work better for non-star topologies or where receiver coverage does not overlap. It gives optimal results if the longest network delay from one member to another is <40 ms and reasonable results if the longest delay is <80 ms.

Switched voting

The 'switched' voting type is selected when channel group receivers have different frequencies. It effectively disables RF voting. There is no 'blending' of streams from different sources. Instead, if more than one call is received on the channel group's RF interfaces, the first call to arrive wins and is provided to the base station's switch. It cannot be outvoted by another signal of better quality. However, if the first call was analog FM and a subsequent call is digital P25, the subsequent call takes over, pre-empting (replacing) the analog FM call.

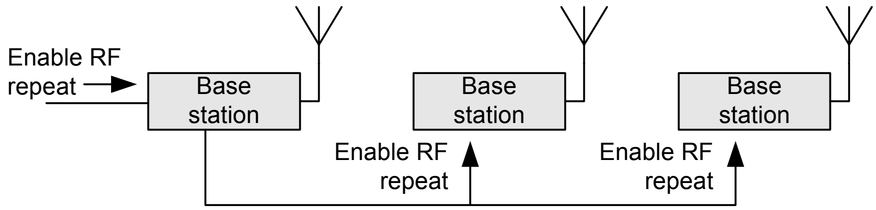
Voting criteria

The following criteria determine the outcome of central or distributed voting:

- If the streams have the same type, the stream with the best signal quality wins the vote. Generally, the impairment is used. This is the inverse of signal quality and its value is included in voice streams. The impairment varies between 0 (best signal quality) and 15 (worst signal quality). The central voting of digital P25 streams uses the number of corrected errors instead of impairment.
- If the streams have the same impairment, the stream at the member with the lowest receiver number wins the vote.

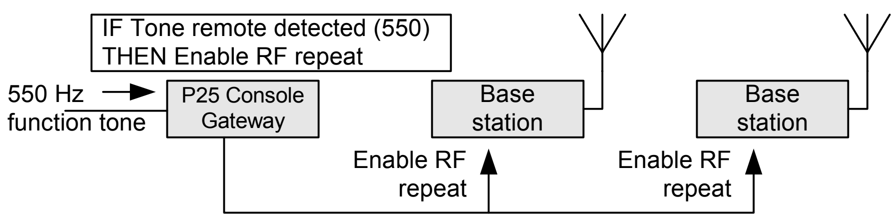
Channel Group Coordination

When a dispatch console is connected to a channel group, dispatch commands such as change channel and enable RF repeat need to be carried out by each member of that channel group. As they are only received by the P25 Console Gateway or by a base station with an analog line or DFSI, there needs to be a process that passes on these commands to the other channel group members. This process is carried out by the channel group coordinator.



The channel control coordinator is a software process inside the channel module of each channel group member. The channel control coordinators work together to implement channel control commands and to ensure that the states of all members are consistent. If collective channel control is enabled, a channel control coordinator sends any commands it receives to other members. Collective channel control is generally enabled when network elements belong to a channel group. See "Collective Channel Control" on page 35. Using the CSS, you can monitor the coordinator of the connected network element.

If the dispatch console is connected via an analog line, the connected network element needs a Task Manager task that interprets the function tone signaling.



Channel control commands

The following table gives an overview of the commands for which collective channel control is available. Note that the functions that some commands invoke can also be obtained through configuration.

Command	Function Selectable through Configuration?	Can be Actioned by Task Manager?	Collective Control
Select channel number	No	Yes	Selectable
Enable/Disable repeat	Yes	Yes	Selectable
Select/Disable receiver	No	No	Always
Monitor squelch	Yes	Yes	Selectable

Channel control messaging

Channel control messaging is handled by the channel group coordinator in each channel group member. When a member with a DFSI or analog line receives a channel control command, the coordinator acknowledges the command and multicasts it to the channel group. The coordinator also resolves any contention (for example if there is more than one dispatcher connected to the channel group) and ensures that members maintain a consistent state. The coordinators can handle members entering or leaving the channel group because of power failure, channel change, or other uncontrolled activities.

2.4 Channel Group Configuration Options

The operation of a channel group can be modified in various ways by changing configuration settings using the CSS. It is important that these settings are consistent across all members of the channel group. The following settings are relevant.

Network Access Codes

Every P25 transmission contains a network access code (NAC). Transmitters must be configured with a transmit NAC. Receivers can be configured to only unmute to a particular NAC or to unmute to any NAC.

The channel group responds to NACs in a similar way to an individual repeater. However, channel groups can also receive inputs from an analog line and/or a DFSI. If the channel group is transmitting signals originating from an analog line, it always uses the configured transmit NAC. If the signals originate from the DFSI and the channel group is configured to transmit the NAC from the voice stream, the channel group transmits using the NAC provided over the DFSI. The channel group also makes the received NAC available on the DFSI.

NACs are configured in signaling profiles. You can enter a receive and a transmit NAC. The ‘Accept any’ check box overrides the receive NAC and causes the receiver to unmute to any NAC. The ‘From stream’ check box affects the transmitter and causes it to use the NAC in the voice stream in preference to the configured transmit NAC.

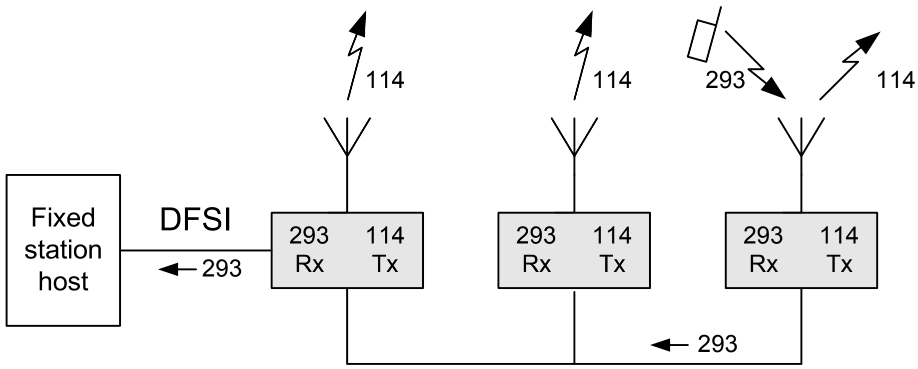
Note Task Manager can configure base station behavior based on the NAC received. You can use this for particular applications, such as a selective cross-band repeater for VHF/800 MHz. When the VHF receiver receives a particular NAC, it changes channel so that it belongs to a different channel group, which links it to the 800 MHz repeater.

You can configure a channel group for the following types of operation:

Single user group

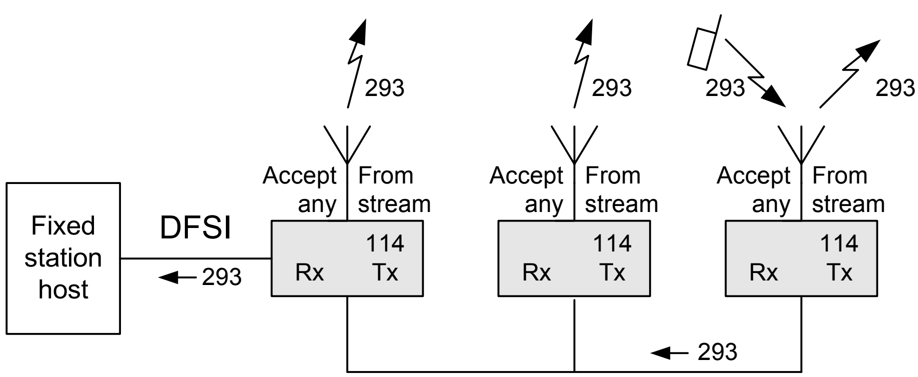
If the channel group is used by a single organization, SUs can transmit using one specific NAC. Base stations also transmit using another specific NAC. SUs only unmute to transmissions with the base station NAC and base stations only unmute to transmissions with the SU NAC. The NAC is not used for selective squelch (i.e. to identify a talkgroup).

Normally, base stations are given the default NAC 0x293, unless there are other P25 base stations nearby that are using the same frequency. The 'Accept any' and 'From stream' check boxes are cleared.

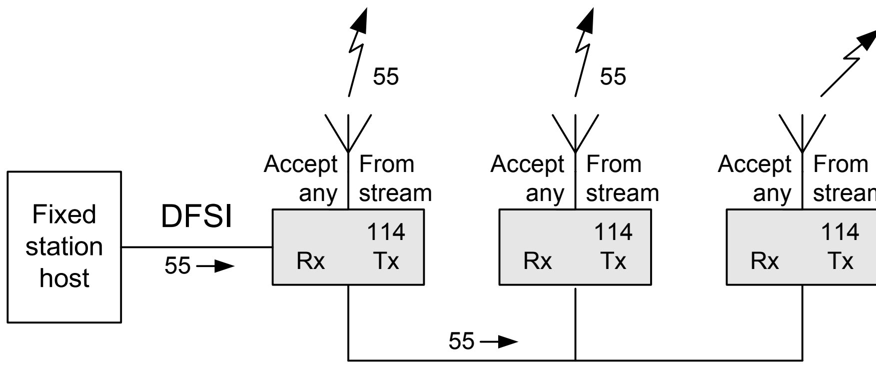


Community repeater

If the channel group is used by different organizations, base stations can accept any NAC and re-transmit it. SUs transmit using the NAC for their organization. For this application, the 'Accept any' and 'From stream' check boxes should be selected. This tells the receiver to unmute to any NAC and also to transmit using the received NAC when the voice stream comes from the channel group (other base station).

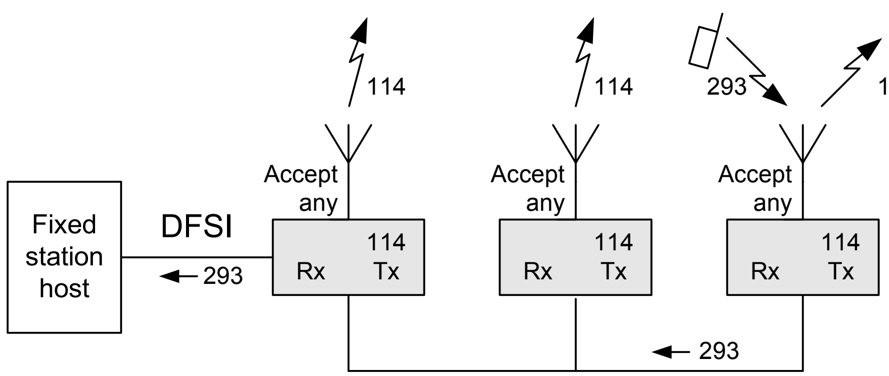


If the digital dispatch console (FSH) provides a NAC, the 'From stream' configuration ensures that the channel group uses that NAC when transmitting.



Hybrid operation

If base station transmitters from other organizations are causing interference, the channel group can be set up to transmit its own NAC but to receive any NAC. SUs are programmed to only unmute to the channel group NAC, so they ignore signals from transmitters belonging to other organizations.



Special receive NAC values

The APCO P25 standards specify two special receive NAC values. F7F lets a repeater receive any NAC and transmit the received NAC. F7E also lets a repeater receive any NAC but it transmits its own configured NAC. Tait has implemented these special values so that the receiver and the transmitter configuration are completely de-coupled.

- To configure a TB9100 for F7F operation, select the 'Accept any' and the 'From stream' check boxes.
- To configure a TB9100 for F7E operation, select the 'Accept any' check box.

Simplex/Duplex

The RF interface of a TB9100 base station is always duplex: it can transmit and receive at the same time. However, the channel group can be configured to be simplex or duplex. This refers to the IP links between channel group members.

A simplex channel group can only handle one call at a time. It selects the one call with the highest priority. Dispatcher calls always win over subscriber calls. If the dispatcher is talking, he or she cannot hear a SU call.

A duplex channel group can simultaneously handle an inbound (radio to dispatcher) and an outbound (dispatcher to radio) call. A dispatcher in a call can hear a subscriber call. Also, one dispatcher can hear another dispatcher and one SU user can talk and hear another SU user (provided voting is disabled and the SUs are duplex).

You might expect that a simplex channel group only handles one voice stream at once, and that a duplex channel group handles two voice streams, one in each direction. In fact, voting and switching means that it is more complex than this. For example, a base station with an analog line can simultaneously send both an inbound and an outbound stream to the channel group. For more details about the number of voice streams that the linking infrastructure needs to support, see "Linking Capacity" on page 50.

Voting

RF voting (described above, see "Voting" on page 28) is configured in the Channel Group dialog box (select Configure > Channel Groups > Channel Group and click Edit.),

To enable distributed voting, configure all channel group members with the voting type Distributed.

If the base station receivers in the channel group use different frequencies, select the voting type Switched.

To enable centralized voting

1. Choose two channel group members (one as the central voter and the other as the backup central voter).
2. Obtain central voter feature licenses for the two members.
3. Configure them with the voting type Central. Make sure that the backup central voter has a higher receiver number than the central voter.
4. Configure the other members with the voting type Distributed. These members must be version 3.05 or above.

When the channel group begins operating, it starts voting in a distributed way. The central voter negotiates with the other members. Once it has determined that there are no problems, it begins functioning as the central voter. The other members cease distributed voting and are now referred to as satellite voters. If the central voter fails, the backup member takes over as the central voter.

Note Digital dispatch equipment can override the channel group's voting procedure and select a vote-winning receiver or disable any receiver. See "Voter control" on page 39.

RF Repeat

If RF repeat is enabled in a standalone base station, the transmitter retransmits signals received on the RF interface.

If RF repeat is enabled in the members of a channel group, the transmitters repeat the inbound RF signal. This could come from any receiver in the channel group. If voting is enabled, the channel group votes and provides the best signal for repeating and sending to other interfaces. If voting is disabled, the voice stream with the highest priority is selected.

RF repeat is configured in the channel table. It can be enabled, disabled, or placed under dispatcher control. For digital dispatcher control of RF repeat, see "RF repeat" on page 39. For analog dispatcher control of RF repeat, see "RF repeat" on page 44.

If RF repeat is dispatch-controlled, dispatcher commands can enable or disable it. If the dispatch equipment is connected to a channel group, members need to be configured to put RF repeat under collective control (see below). Then the dispatcher commands are passed on to all channel group members and do not only affect the connected member.

An RF repeat command from the FSH directly turns RF repeat on or off. By contrast, analog dispatch equipment can only send particular tone remote function tones. These require Task Manager actions that turn RF repeat on or off in response to the function tones.

Collective Channel Control

Collective channel control is generally enabled in channel groups with a dispatch interface. It is enabled separately for channel change, enable/disable RF repeat, and monitor commands. Other commands (wildcard commands) can be propagated to the channel group using Task Manager.

Enabling collective channel control

To enable collective channel control

1. In the channel group configuration of each member (Configure > Channel Group > Channel Groups and click Edit), set the channel control of the required dispatch commands to 'collective.'

2. In the channel table of each channel group member, set RF repeat to 'DispatchControlled.'

3. If the dispatcher interface is analog, set up Task Manager actions at the P25 Console Gateway that implement the dispatch command on receiving the relevant function tone. (It is necessary to implement the command at the Console Gateway even though this network element cannot repeat or monitor RF, so that collective control can pass the command on to the other channel group members.)

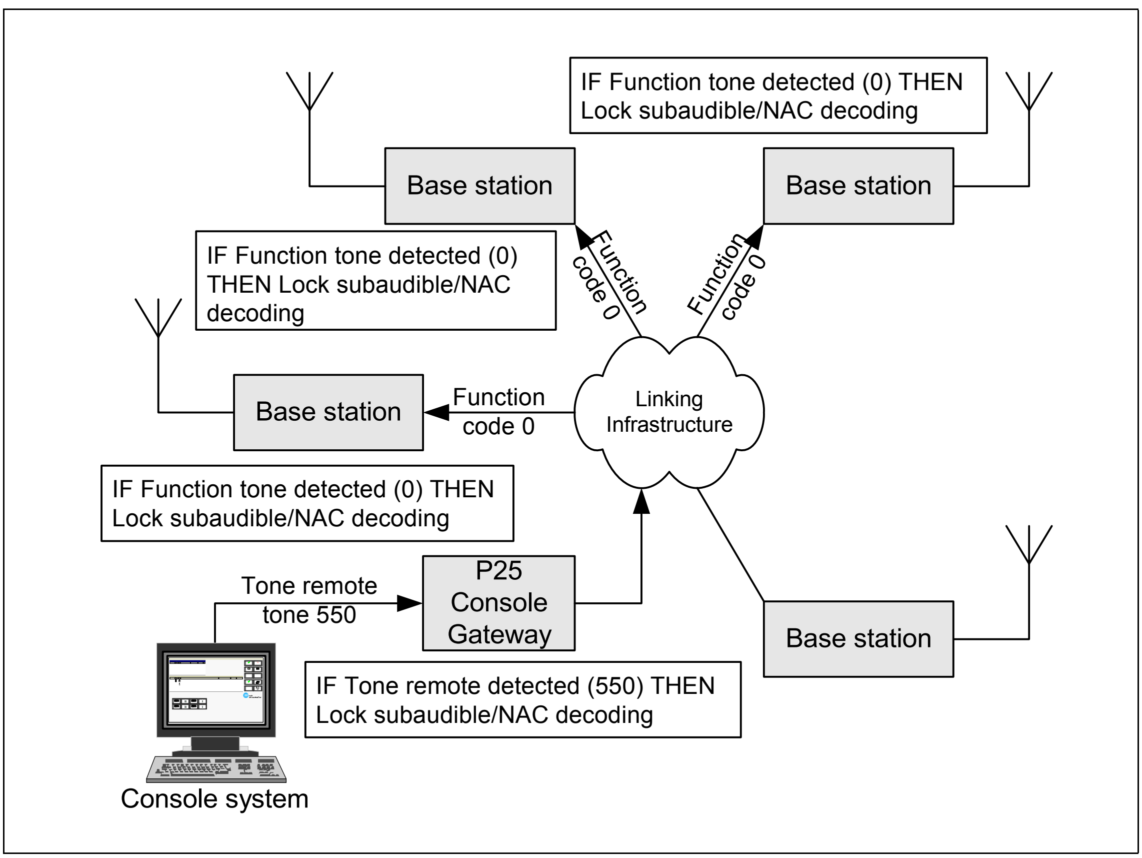
Alternative method — manual implementation using Task Manager

Collective control of the channel group can alternatively be obtained by hand-coding the propagation to other channel group members. This is done using the 'Send function code' Task Manager action. This can be used for wildcard channel control commands that are not subject to collective channel control. Please note that there is no monitoring of control status so that, if a member is out of service when the wildcard is propagated, it will be out of step with the rest of the channel group.

Important Make sure that you do not use Task Manager tasks involving function codes for channel change, RF repeat, or Monitor commands if they are under collective control.

Figure 2.1 shows how this alternative method works.

Figure 2.1 Propagating a dispatcher wildcard command



To propagate a 'Lock subaudible/NAC decoding' command

1. In the connected base station or P25 Console Gateway, set up the Task Manager statement:
   IF ***Tone remote detected (550)*** THEN ***Lock subaudible/NAC decoding***
   IF ***Tone remote detected (550)*** THEN ***Send function code 0***

Note You can select any function code between 0 and 255 as the equivalent of a tone remote function tone.

2. In each of the other base stations in the channel group, set up the following Task Manager statement:
   IF ***Function code received (0)*** THEN ***Lock subaudible/NAC decoding***.
   When they receive the function code, they also lock the decoding of subaudible signaling and NACs, so that they unmute to any signal.

Jitter Buffer Settings

Each TB9100 base station has a configurable transmit buffer. The buffer is needed to accommodate delay variation when sending voice streams across the network. Two configurable timers determine the size of the transmit buffer: the transmit holdoff and the preamble. While the jitter buffer is filling, the base station waits (sometimes) for the transmit holdoff time and then transmits a preamble for the time defined by the preamble timer.

The following offers general guidance on the settings to choose for different systems.

Systems without SU scanning or voting

In this type of channel group, the main consideration is that the jitter buffer is large enough to prevent buffer underruns and the consequent drop-out of speech. In most situations, setting the **Preamble** to 40 ms (for jitter originating in the base station and its interface) plus an allowance for jitter originating in the routers should be sufficient. Otherwise follow these steps.

1. Using the CSS (Monitor > Interfaces > Channel Group), monitor the quality of service of incoming calls and note the jitter.
2. Select Configure > Channel Group > Network.
3. Under **Jitter Buffer**, enter into the **Preamble** box the jitter value you noted plus 40 ms.
4. Monitor the transmitter jitter buffer (Monitor > Interfaces > Channel Group.) If underflows occur, increase the preamble length.

The **Transmit holdoff** can be set to 0.

Systems with SU (downlink) voting and central uplink voting

In networks with downlink voting, subscriber units scan the available channels and vote on the best signal. Set the **Preamble** to a long enough value (for example, 150 ms) to allow the radios to scan the channels and vote on the best signal before the call itself begins (avoiding late entry is particularly important with encrypted speech). Set the **Transmit hold-off** to 0 at the base station with the longest network delay from the central voter. Calculate a transmit holdoff value for each other transmitter by subtracting its network delay from the longest network delay. Base stations in centrally voted channel groups always apply their transmit holdoff.

Systems with SU (downlink) voting and distributed uplink voting

In networks with downlink voting, subscriber units scan the available channels and vote on the best signal. Set the **Preamble** to a long enough value (for example, 150 ms) to allow the radios to scan the channels and vote on the best signal before the call itself begins (avoiding late entry is particularly important with encrypted speech). The **Transmit hold-off** should be the (average) network delay. It is only applied by the base station receiving the call and delays the call start until the other base stations are ready to transmit.

Systems with SU scanning

In networks with downlink scanning, subscriber units scan channels until they find a signal. Once a signal is detected, scanning ceases. Set the **Preamble** to a sufficiently long value to allow the radios to scan the channels in the list. The **Transmit holdoff** can be set to 0, as there is no SU voting.

2.5 Digital Fixed Station Interface

Any base station or P25 Console Gateway can provide its channel group with a digital fixed station interface (DFSI). A DFSI feature license is required. A channel group can have more than one DFSI.

Channel Control

The DFSI supports the following dispatcher commands for channel control. Most commands need collective channel control (see "Collective Channel Control" on page 35) to coordinate them across a whole channel group.

Dispatcher Command	Propagation of Command
Monitor	Collective setting
RF repeat enable/disable	Collective setting
Change channel	Collective setting
Voter control (select or disable receiver)	Member with DFSI forwards command to channel group

Monitor

(Not supported in Version 3.0) The dispatcher can send a monitor command in order to listen to the channel group. This disables checking for NAC, destination ID, and subaudible signaling. When the channel group receives a monitor command, any calls that it receives are forwarded to the FSH, even if that call has the wrong NAC, destination ID, CTCSS tone or DCS code.

RF repeat

The dispatcher can enable or disable RF repeat. Channels in the channel table must be configured to put RF repeat under dispatcher control. If channel control is collective for RF repeat, the dispatcher command is implemented in all members of the channel group.

To set up commands for enabling and disabling RF repeat

1. In the channel table of the member with the DFSI, set the **RF repeat** column to DispatchControlled.
2. For all channel group members, configure the channel group settings to have collective control of RF repeat. (Configure > Channel Group > Channel Groups and click Edit.) This ensures that the command is automatically coordinated across the channel group.

Channel change

The dispatcher can instruct the channel group to change to a particular channel number. If channel control is collective for this command, all members of the channel group change to that channel.

Voter control

The dispatcher can exercise control over the voting decisions of the channel group. If the dispatcher selects a particular receiver number, that receiver's voice stream always wins the vote. The other receivers are disabled. However, if the selected receiver is not receiving a call, channel group members can provide voice streams from their control panel or from another dispatcher interface. Alternatively, the dispatcher can send a command to disable a particular receiver. It then does not provide its RF voice stream to the voter(s).

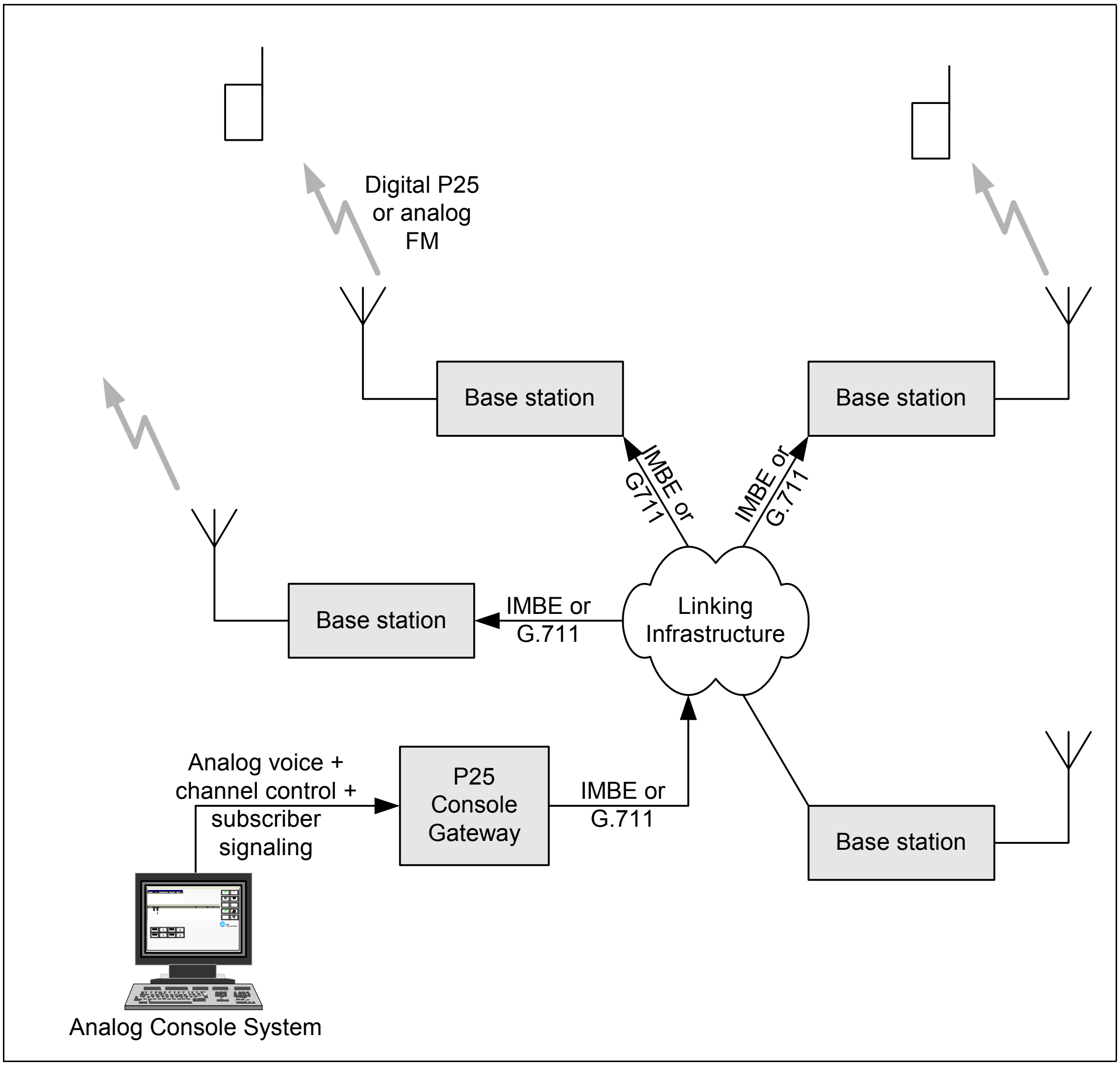
Subscriber signaling

Digital dispatch equipment is able to provide subscriber signaling (individual ID, caller ID, group ID) over the DFSI. This signaling is embedded in the RTP voice stream. Digital dispatch equipment is also able to send and receive TSBKs.

2.6 Analog Line Interface

An analog line provides the interface between a channel group and analog dispatch equipment. It can also interface to an analog FM base station or to call recording equipment. This interface is normally provided by a P25 Console Gateway, but if encryption/decryption is not required, it can be provided by a TB9100 base station. The base station needs an analog line feature license.

Figure 2.2 Interfacing a channel group to a console system



Channel control

Dispatcher commands can exert channel control over the channel group. Control is exerted over all members of the channel group, not just to the member with the interface. Where needed, this is done through a 'collective' setting. The following table summarizes the supported analog dispatcher commands and indicates the signaling methods that dispatch equipment can use to communicate these commands over the interface. It also indicates whether the command needs to be propagated to all channel group members and whether a collective setting is available to enable this.

Dispatcher Command	Available Signaling Methods				Propagation of Command
	E & M	Keytone (LLGT)	Function Tone (Selecting a Calling Profile)	Function Tone (Triggering a TM Task)	
Tx Key (channel seize)	Yes	Yes			Not needed
Rx Gate (analog valid)	Yes				Not needed
Select mode (analog FM/ digital P25)			Yes		Not needed
Monitor			Yes		Collective setting
Select encrypted/clear			Yes		Not needed
RF repeat enable/disable			Yes		Collective setting
Change channel				Yes	Collective setting
Wildcard (trigger TM action)				Yes	Function codes

Subsequent sections describe the signaling methods and dispatcher commands used for channel control and indicate how they are enabled and configured using the CSS.

E & M signaling

In a traditional analog system, the E & M lines are used for Rx Gate and Tx Key signals. Rx Gate tells the console system that the receiver has unmuted. Tx Key tells the base station to transmit.

In a channel group, things are more complicated. An Rx Gate signal does not necessarily mean that the receiver has unmuted. It does indicate that there is a valid audio signal, but this might have been received by any channel group member or have come from the control panel microphone. Similarly, a Tx Key signal doesn't necessarily key the transmitter, as any incoming signal competes for selection with other signals. Accordingly, at the analog line interface, we use the terms 'Analog valid' instead of Rx Gate and 'Channel seize' instead of Tx Key.

Analog valid indicates that there has been a selection and that it has been switched to the analog line out. Channel seize indicates that there is a signal on the analog line and asks the channel group to consider it for selection.

To configure the E & M Lines for Analog valid and Channel seize

1. Select Configure > Analog Line > General.
2. Under **Channel seize and analog valid**, select the **E & M** check box.

Keytone

Keytone (also known as LLGT–low level guard tone) can convey a channel seize signal. This instructs the channel group member to present the audio from the analog line to the channel group for selection. If the audio is selected, it is transmitted by the base stations in the channel group. When the keytone stops, the channel group stops transmitting.

To configure the detection of keytone

1. Select Configure > Analog Line > Tone Remote Options.
2. In the **Guard tone frequency** box, select the keytone frequency.
3. In the **LLGT level** box, select a minimum level for the keytone. In the **HLGT level** box, make sure that the level selected for high level guard tone exceeds the actual keytone level.
4. Enable the notch filter.

To configure keytone as a channel seize signal

1. Select Configure > Analog Line > General.
2. Under **Channel seize and analog valid**, select **Tone remote**.
3. If MDC1200 signaling is used, select **After MDC1200**. This configures the channel seize to occur some time after LLGT begins, so that MDC1200 signaling is not included in the transmitted voice stream.

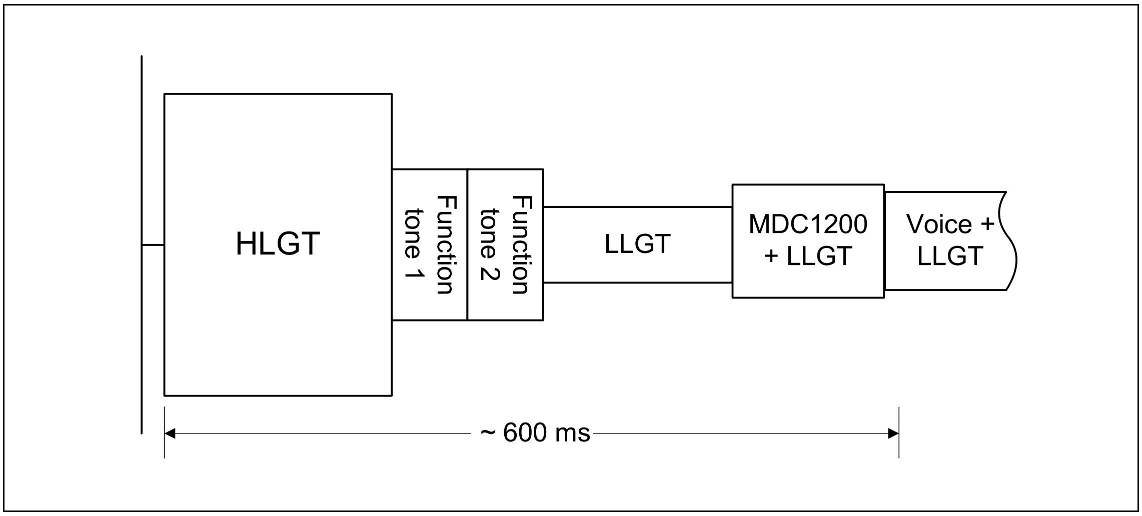
Tone remote function tones

The analog line supports the use of single and dual-tone tone remote signaling from console to channel group member. The tones are carried on the audio line in. The channel group member can respond to a tone remote function tone by selecting a calling profile or by carrying out Task Manager actions with that function tone as an input. Selecting a calling profile overrides the calling profile assigned to the current channel.

Note Subscriber-related settings in the calling profile can be overridden by MDC1200 signaling.

Tone remote signaling consists of a high-level guard tone (HLGT), followed by one or two function tones, then low-level guard tone (LLGT), accompanied by audio.

Figure 2.3 Tone remote signaling



To configure tone remote signaling

1. Adjust the parameters for detecting tone remote, so that they conform with the tones that the console system produces.
2. Create calling profiles and assign them to the tones that the console system will use.
3. For commands that cannot be implemented using calling profiles, create channels in the channel table and use Task Manager tasks to select them when the appropriate tone is received.

Select mode

If the channel group supports both analog FM and digital P25 users, the dispatcher needs to be able to select the mode when initiating a call. Configuring the base station for Select mode commands is done by defining calling profiles, each with the required mode, then assigning function tones to those profiles.

Selecting a calling profile not only assigns the mode, but many other properties of the analog line as well. This means that the dispatcher may need several different calling profiles (you can define up to 16).

To set up commands for selecting analog FM or digital P25 modes

1. Set up calling profiles for the different modes: one for Analog FM and one for each group or individual that the dispatcher wants to call in digital P25 mode.
2. Assign the calling profile to be used as the default to the channel or channels that the base station will normally use
3. Assign the calling profiles to the function tones that the console system will use (Configure > Analog Line > Tone Remote Mapping)

4. Make sure that the channel profiles assigned to the channel(s) that the base station uses support receiving both analog FM and digital P25 (otherwise the dispatcher will be able to talk to the radios but the base station will not be able to receive their reply).

5. Assign the function tones to suitable controls on the console system. Activating these controls will select the corresponding calling profile. Assigning a function tone that changes calling profile to the PTT button is not recommended, as the call cannot begin until the profile has been changed.

Note When the console transmits in one mode, it can still receive calls in the other mode. To reply, the dispatcher must first select the mode used by the caller.

Monitor

A Monitor command instructs the base stations in the channel group to defeat NAC, CTCSS, and DCS. (Not supported in Version 3.0x)

RF repeat

The dispatcher can enable and disable the RF repeat function of the channel group.

To set up commands for enabling and disabling RF repeat

1. In the channel table of the member with the analog line interface, set the **RF repeat** column to DispatchControlled.

2. Set up Task Manager tasks such as the following:
   IF ***Tone remote detected (550)*** THEN ***Enable RF repeat***
   IF ***Tone remote detected (750)*** THEN ***Disable RF repeat***

   Even though a P25 Console Gateway cannot repeat, you need to provide it with these tasks for sending to members who can.

3. For all channel group members, configure the channel group settings to have collective control of RF repeat. (Configure > Channel Group > Channel Groups and click Edit.) This ensures that the command is automatically sent to all channel group members.

Change channel

If the dispatcher needs to communicate with users on a different frequency pair, or to change other operating parameters of the base stations in the channel group, the base stations can be configured to respond to a dispatcher command by changing channel. This is done through Task Manager.

To set up a command for changing channel

1. In the P25 Console Gateway, create a Task Manager statement with the function tone as input and a change channel command as action.

   For example: IF ***Tone remote detected (550)*** THEN ***Go to channel 5***.

2. Configure all channel group members to have collective control of the channel number (see “Collective Channel Control” on page 35).

Note The dispatcher should not use the change channel command when the base station is transmitting. The change to the new channel's calling profile will not take place until the analog line initiates a new transmission.

Subscriber Signaling

Subscriber signaling is passed over the analog line mostly as MDC1200. However, function tones can be used to select calling profiles for digital P25 calls that specify an individual or group address and a sender ID.

The analog line supports the use of MDC1200 (also known as Stat Alert) for subscriber signaling.

In analog FM mode, the channel group passes MDC1200 transparently between dispatch console and radio and also between radios. No configuration is required.

In P25 digital mode, the analog line can convert MDC1200 signaling to P25 signaling and vice versa. This enables analog console systems to make use of P25 features that are unavailable to tone remote signaling. The other base stations in the channel group do not need to be configured; the P25 signaling is propagated as part of the voice stream.

The channel group member with the analog line requires the following:

- MDC1200 feature license
- MDC1200 signaling must be enabled (Configure > Analog Line > General. Select the MDC1200 check box)
- Each required MDC1200 feature must be enabled
- MDC1200 addresses must be mapped to P25 IDs. Generally, the default mapping is sufficient

Many MDC1200 messages require an acknowledgement and can be retried until an acknowledgement is received. The number of retries is determined by the console system. The retry rate, also determined by the console system, may need to be reduced to take account of the slower response when the message and its response need to be converted to and from digital P25.

The following sections describe the MDC1200 signaling messages that are supported in digital P25 mode. They also provide configuration instructions.

Caller identification (ANI)

The most common use of MDC1200 signaling is to provide the called party with the identity of the caller. In analog systems, radio equipment can send a MDC1200 message containing the ID of the caller. This ID can be displayed on the receiving radio or console system. The feature is known as ANI (automatic number identification) or PTT ID. Console systems can also send ANI messages.

In digital P25 mode, the TB9100 can take the source ID of an incoming P25 call, convert it into a MDC1200 ANI message with an equivalent MDC1200 address, and send it to the dispatch console. For outgoing calls, the analog line uses the line ID specified by its current calling profile as its ANI.

To configure the analog line for caller identification on incoming calls

1. If possible, set up the analog line for trailing ANI (Configure > Analog Line > General). This adds the ANI message at the end of the over, reducing the delay at the start of the over).

2. If desired, modify the default mapping of P25 addresses to MDC1200 addresses (Configure > Analog Line > MDC1200 Address Table). This mapping converts the source addresses of digital P25 calls it receives into equivalent MDC1200 ANIs.

To configure the dispatcher's caller identification

1. Assign the the dispatch console an MDC1200 address in the range 1–999.

2. Enter the same number into the **Line ID** box of all the analog line's calling profiles. (This assumes the default mapping in the MDC1200 Address Table. You are entering a decimal P25 address that the address table converts into the console's hexadecimal MDC1200 address.) The line ID will be used as the console's caller identification on outgoing digital P25 calls.

Note Many radios will send status messages to a dispatcher ID that is derived from the talk group they currently belong to. You don't need to configure the analog line for these messages. The base station automatically switches them onto the analog line, if the current group membership includes the talk group. The MDC1200 destination ID is calculated from the calling profile's Line ID and not from the actual destination ID of the status messages.

Emergency ANI

This is similar to ANI, except that the message format is slightly different and the associated call is an emergency call.

Outbound calls: When the analog line receives a MDC1200 emergency ANI message, it makes the following call an emergency call (overriding the setting in the current calling profile).

Inbound calls: When the channel group receives a P25 emergency call, the analog line produces an MDC1200 emergency ANI message.

Additional services

MDC1200 can also be used to provide a range of services additional to ordinary voice calls. If these are to be converted to or from their digital P25 equivalent, the current service profile must enable them. Messages on the analog line (in MDC1200 format) must be enabled in the service profile attached to the calling profile. Messages arriving from a receiver in the channel group must be enabled in the service profile attached to the current channel of the base station receiving the message.

Call alert

The dispatcher can send a call alert message to a radio. The radio acknowledges the message and may provide a tone or LED indication that the user is to call back. The radio sends a call alert to the dispatcher, if the service profile assigned to the current channel enables it. The base station can convert a MDC1200 call alert message into a P25 call alert supplementary service and vice versa.

Radio check

The radio check message lets the dispatcher test whether a radio is powered up and within coverage.

Emergency alarm

The dispatcher can receive an emergency alarm message from radios. This is also known as emergency alert or man down. Receiving this message may cause an alarm indication on the console system.

No base station configuration is necessary for this message; it is not possible to disable this function.

Radio disable/radio enable

The dispatcher can disable or enable radios (P25 refers to this as inhibit or uninhibit), provided the radios support the feature and have it enabled.

Remote monitor

The dispatcher can remotely monitor P25 radios on the network. The console system sends the MDC1200 message, which is converted to the equivalent P25 message. The individual address of the radio is converted from MDC1200 to P25 format.

Status update and status request

The dispatcher can request status information from a radio and the radio can send an update on its status.

Call addressing (voice alert)

MDC1200 voice alert messages have been used in analog systems for some time to address calls to particular individual radios or groups. The analog line can turn this into a P25 individual call to the equivalent P25 individual or group, overriding the setting in the calling profile. The default mapping between MDC1200 and digital P25 addresses can be displayed in the CSS and suffices for most purposes.

Function Tone Subscriber Signaling

Where MDC1200 is not available, a dispatcher can use function tones to provide a limited capability for subscriber signaling in digital P25 mode. The function tones select a calling profile that specifies a destination group or individual ID.

Calling groups

To set up a command for calling a group

1. Set up a calling profile and select the call type ‘P25 group’. Enter the group ID in the Destination field.
2. Set up a group membership with that group included. This ensures that the dispatcher can listen to the group.
3. Select Configure > Analog Line > Tone Remote Mapping and assign the calling profile you created to the function tone that the console produces when the button is pressed.

Calling individuals

There is a limited ability to use tone remote signaling to make calls to an individual ID. A calling profile is needed for each individual ID and only 16 calling profiles are available for individual and group calling.

To set up a command for calling an individual

1. Set up a calling profile and select the call type ‘P25 individual.’ Enter the individual ID in the Destination field. Note: the analog line will automatically listen to calls addressed to the Line ID in the calling profile.
2. Select Configure > Analog Line > Tone Remote Mapping and assign the calling profile you created to the function tone that the console produces when the button is pressed.

3 Network Design

The following describes various aspects of the design of conventional TaitNet P25 networks:

- Elements comprising the network
- Network topology
- Determining the required linking capacity
- Using TB9100s for RF linking
- Connecting a TaitNet P25 network to the organization's own network via a firewall
- Enabling remote access and support by Tait engineers
- Making the transition from an existing analog system

3.1 Network Elements

A TaitNet P25 network consists of various network elements: Tait products (the TB9100 base station and the P25 Console Gateway) and off-the-shelf network equipment (switches, routers, and hubs). Only Tait-approved equipment should be used. It must support particular features such as multicast, VPN, proxy ARP, and IGMP. Trunked networks also have one or more trunking site controllers.

The network can interface to third-party dispatch equipment and voice recorders.

The Tait network elements are modular. Different models consist of various combinations of modules. For more information about module combinations, see the TB9100 Installation and Operation Manual. In addition, there is a variety of optional features, each requiring a software feature license.

All Tait network elements have at least one channel module. For a base station, this is a reciter (an RF receiver/exciter). For a P25 Console Gateway, this is a gateway module.

TB9100 Base Station

The TB9100 base station is available as single-channel and dual-channel subracks. Receive-only base stations can have up to five receive channels (or seven receive channels, if there is no power management unit) in a single subrack. TB9100 base stations have many capabilities that require enabling by feature license.

P25 Console Gateway

P25 Console Gateways have a digital line (IP over Ethernet to connect to the network) and a 4-wire E & M analog line to connect to the console equipment. Their encryption capabilities are similar to Tait P25 mobiles and portables. They can store up to 16 encryption keys. The DES and AES algorithms are supported.

The P25 Console Gateway has a standard P25 key-fill interface for loading encryption keys. This can be done using the Motorola KVL3000+ key fill device. P25 Console Gateway operation can be monitored and configured using the same CSS as is used for the TB9100 base station.

Currently, the P25 Console Gateway consists of a TB9100 reciter with the receive and transmit functions disabled.

3.2 Topology

TaitNet P25 networks normally use a star topology. This means that there is a maximum of two hops between any two members of 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.

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 provide alternative data routes, which may result in packets being received out of order. Base stations drop these packets and substitute empty values for them, resulting in a loss of audio.

3.3 Linking Capacity

Once the network topology has been decided upon, the bearer network needs to be provisioned with sufficient IP capacity to reliably carry voice traffic, even in the presence of other IP data. This section identifies the variables that affect the load on the network and gives some rules of thumb to assist network designers to adequately estimate a suitable linking capacity. The end result will involve a trade-off between cost and quality of service.

Variables

The following variables affect the loading on the network.

Calling types

P25 calls require less network bandwidth than FM calls. P25 calls use an IMBE vocoder to reduce the raw speech bit rate. The vocoder data rate is 4.8 kbit/s. With the addition of P25 framing information, IP network framing overhead and voting information, the data rate increases to approximately 15 kbit/s. FM calls are converted to the G.711 format for transmission on the network. The basic data rate of G.711 is 64 kbit/s and it therefore requires significantly more linking capacity.

Voting type

If there is no common uplink frequency, receivers are assumed not to be listening to the same signal and the channel group is configured with the voting type 'switched.' If two SUs begin overs at the same time, there is a single transient load spike at the beginning of the over as the receivers choose which SU to handle. This single load spike can be disregarded when estimating linking capacity.

If voting is centralized, voice streams are sent from multiple receivers to the central voter. The capacity requirement is well-defined.

If voting is distributed, several voice streams can be on the network simultaneously, causing transient load spikes throughout the over. Although most of the time the load is low, linking capacity must be sufficient to handle these spikes. Estimating this capacity is more difficult.

Simplex/duplex channel group

A channel group can be simplex or duplex. When the channel group is duplex, each member votes separately on two voice streams: inbound (from RF to the dispatcher) and outbound (from the dispatcher to RF). A base station in a duplex channel group is capable of sending one stream and receiving another stream over the digital line at the same time. However, if it also has a dispatcher interface, it must be capable of sending two streams simultaneously; one from its RF interface and the other from the dispatcher interface.

Network delay

Increasing the network delay also increases the load on network links in systems with distributed voting. The network design should aim to minimize the network delay.

Network topology

Having more than two hops in a network adds to the network delay. Tait strongly recommends using a Star topology. For centralized voting, the star topology is necessary. The figures and calculations below are based on a Star topology, which has no more than two hops.

Linking Options

To keep the linking capacity to an absolute minimum, use routers and configure them with compressed RTP (CRTP). If you use routers, configure them for QoS support. If linking capacity is plentiful, use switches instead of routers. They are easier to set up, operation under load is more predictable, and they minimize network delay.

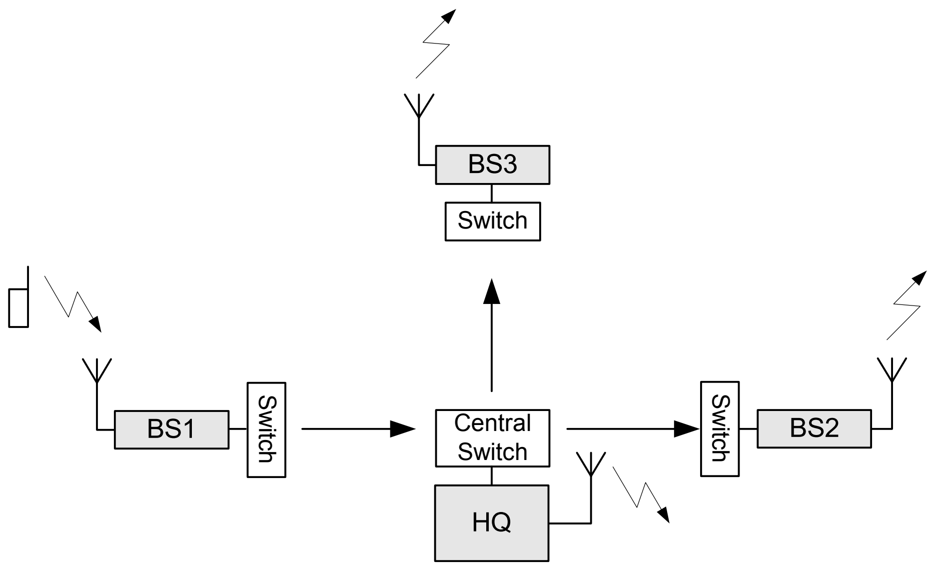
Facility	Description	Advantages	Disadvantages
Compressed RTP	Compress the IP packets containing voice	Reduces the network load to an absolute minimum	Requires CISCO routers
			Can be complex to set up
			Less predictable
			Does not scale to large networks (> 20 channel groups)
			Cannot be used with links of 2 Mbit/s and beyond
QoS support	Prioritize the real-time (voice) stream ahead of other traffic	Minimal additional delay when other traffic is present	Requires routers
Switches	Use a VLAN without routers. No other traffic shaping required.	Network is simple and predictable	Cannot use QoS and so must over-provision to accommodate non-voice traffic
		Delays are minimized	

Determining the Number of Voice Streams

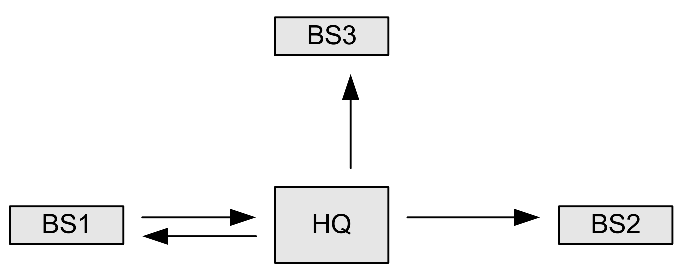
To estimate the required network bandwidth, you need to determine the maximum number of simultaneous voice streams that the links between sites must be able to carry. The number of streams from the central switch or router to the network element (HQ) at the center of the star topology is not of interest, provided that the link is Ethernet (high bandwidth). HQ can be a P25 Console Gateway, a base station, or a base station with a digital dispatcher interface.

The following discussion assumes that the dispatch equipment interfaces to the network element at the center of the star topology. If it interfaces to another network element, that element's link may need to handle more streams.

The following diagram shows the streams that appear on intersite links when a SU makes a call in a channel group with four members.

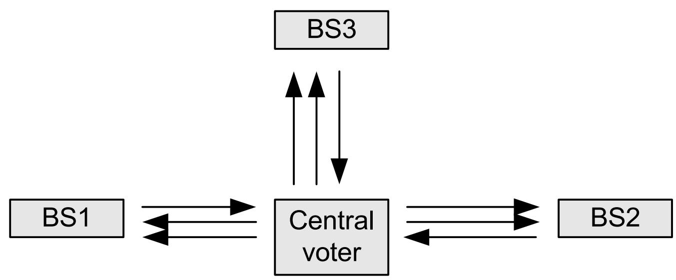


Switched. In systems where the voting type is 'switched', there only needs to be sufficient capacity to handle a single voice stream in any one direction. Any base station (BS1 in the diagram below) can provide a subscriber stream to the network and at the same time receive a dispatcher stream.



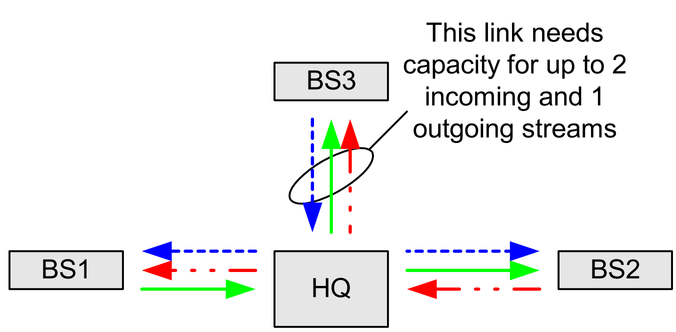
(There is an exception to this rule: In duplex channel groups, if a channel member has a RF and a dispatcher interface, it can have two outgoing voice streams: one from the receiver and the other from the analog line or DFSI.)

Central Voting. When voting is central and the channel group is duplex, links to the central voter can have one incoming and two outgoing streams. (While the Ethernet link from the central switch or router to the central voter carries one incoming stream from each active receiver, each link into the central switch or router only has one incoming stream.)



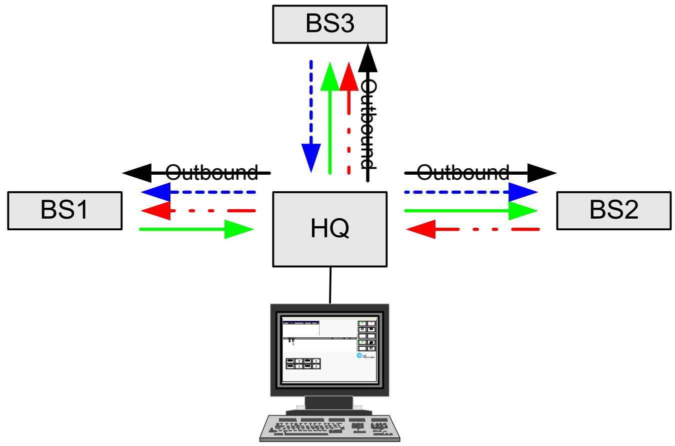
If the channel group is simplex, the central voter only has one outgoing stream. However, if a channel member has a RF and a dispatcher interface, it can send two streams to the central voter.

Distributed Voting. When voting is distributed among the members of the channel group, there are transient load spikes in which there can be more than one voice stream on a link at the same time. Each additional simultaneous receiver adds a voice stream to the outgoing link. In the following example, three base stations in the channel group are simultaneously receiving the same signal.



Note that there is only one outgoing stream, but *n*–1 incoming streams, where *n* is the number of simultaneous receivers. The P25 Console Gateway is an exception, having *n* incoming streams.

If the system is duplex, there is an additional stream originating from the dispatcher:



The following table indicates the number of voice streams involved for different scenarios in voted systems.

Channel Group Type	Link	Number of Voice Streams	
		Two Receivers	Three Receivers
Simplex	Incoming	1	2
	Outgoing	1	1
Duplex	Incoming	2	3
	Outgoing	1	1
	Incoming to P25 Console Gateway[a]	2	3
	Outgoing from P25 Console Gateway[a]	1	1
	Incoming to base station with DFSI[a]	1[b]	2[b]
	Outgoing from base station with DFSI[a]	2	2

a. This is relevant only when the network element is not positioned at the center of the star topology.

b. Assumes that the base station is one of the receivers

Estimating the Link Capacity Requirement

If the linking capacity needs to be kept to a minimum, use the following section to estimate the requirements. This is fairly straightforward for non-voted and centrally voted systems, but more complex for systems with distributed voting.

Step 1: Determine the bandwidth of the voice stream

The following are estimates of the capacity requirement for each type of voice stream.

Call Type	Compressed RTP	RTP
Digital P25	19.2 kbit/s	54 kbit/s
Analog FM	100 kbit/s	120 kbit/s

Step 2: Adjust for the number of voice streams

To estimate the total capacity required by the voice streams

1. Determine the maximum number of voice streams (see "Determining the Number of Voice Streams" on page 52)
2. Multiply the allowance for the voice stream (Step 1) by this number.

Step 3: Add an allowance for administrative traffic

The basic requirement for the voice stream includes an allowance for maintenance traffic (alarms, configuration, monitoring). This is generally sufficient for routed networks. These have QoS, which prioritizes speech traffic.

- Lower-capacity links may need additional bandwidth to handle this traffic.

Switched networks do not have QoS and therefore need an allowance for maintenance traffic, as follows:

Service	Additional allowance
CSS monitoring	23 kbit/s (per CSS)
Syslog messages to collector (Trace level)	16 kbit/s (per base station)
Syslog messages to collector (Notice or Warning level)	5 kbit/s (per base station)

Step 4: Combine the estimates for each channel group

To estimate the total capacity for each mono-directional link:

- Add together the capacity that you estimated separately for each channel group

Example calculations

The following provide two example calculations resulting in capacity requirements for one channel group.

Centrally voted system

The channel group is duplex. The network is routed and without CRTP. Calls are P25 only.

Voice stream base capacity (Step 1)	= 54 kbit/s
Maximum number of voice streams (incoming): (Step 2)	= 1
Maximum number of voice streams (outgoing): (Step 2)	= 2

108 kbit/s is the linking capacity requirement for the channel group for each mono-directional link.

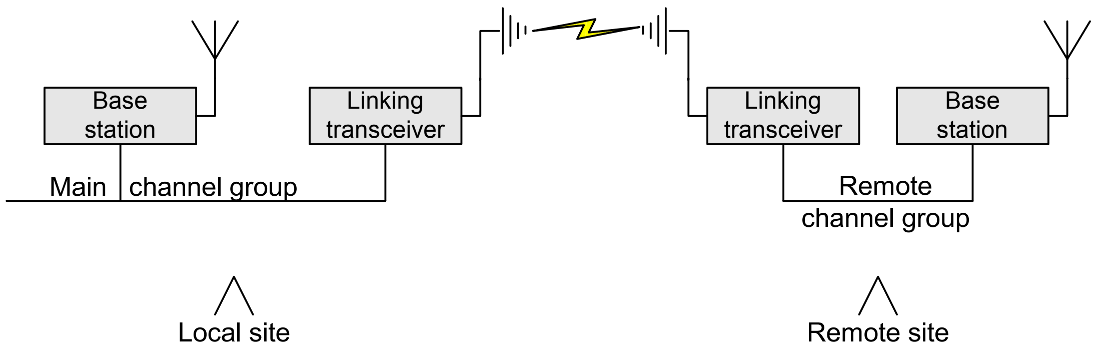
Simplex system with distributed voting

The channel group is simplex. The network is routed and without CRTP. Calls are P25 only. There can be up to three simultaneous receivers.

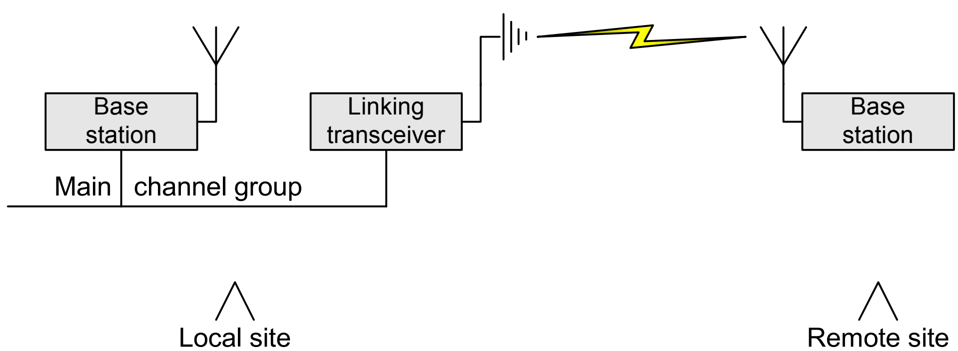
Voice stream base capacity (Step 1)	= 54
Maximum number of voice streams (incoming): (Step 2)	= 2
Maximum number of voice streams (outgoing): (Step 2)	= 1
Allowance for administrative traffic (Step 3)	= 0
Linking capacity requirement (incoming streams):	= 2 x 54 = 108 kbit/s
Linking capacity requirement (outgoing streams):	= 1 x 54 = 54 kbit/s

3.4 RF Linking

An ordinary RF voice channel can be used to link a channel group to a remote TB9100 base station or to a mobile dispatch center. In TaitNet P25 networks, this RF linking is provided by TB9100s especially configured for the task. They are referred to as linking transceivers. RF linking can be a cost-effective solution in areas where an IP-based infrastructure is not present or would be expensive to provide.



It may be possible to use an ordinary TB9100 base station, functioning as a repeater, as one end of the link, with significant savings in equipment cost:



Uses and Limitations

RF linking can be used for the following applications:

- Extending the coverage area of a channel group. RF linking connects a remote base station to the channel group. (The remote base station's coverage could in turn be extended by adding fill-in receivers or even additional base stations.)
- Giving a mobile dispatch console access to a channel group. RF linking connects a DFSI interface or analog line interface to the channel group.

RF linking has the following limitations.

- It cannot be used in analog FM mode.
- It is not suitable for joining channel groups if both of them have line interfaces as well as RF interfaces.
- It performs better with switched or distributed voting than with central voting.
- It can be used in simulcast systems, but the remote channel group cannot be synchronized to the main channel group.
- It may give unpredictable results if a maintainer uses the control panel at the remote location.
- It must use a specific NAC configuration, if one end of the link is an ordinary base station/repeater.

Determining Site Suitability

Each linking transceiver needs to receive a strong enough signal to provide an adequate fade margin. The TB9100 has the following P25 static sensitivity values (see also the TB9100 / P25 CG Specifications Manual). Use them to determine the suitability of the site locations and equipment for an RF link.

Signal Strength	Sensitivity Degradation	BER	DAQ
-120.5 dBm	0 dB	5.0%	
-119 dBm	1.5 dB	2.6%	3.0
-118.5 dBm	2.0 dB	2.0%	3.4
-117.5 dBm	3.0 dB	1.0%	4.0

Configuration Overview

The following lists the main configuration tasks needed for a pair of linking transceivers. It assumes that the main channel group has a dispatcher interface. Other applications and system designs need different settings. The remote base station should be configured as an ordinary repeater/channel group member.

1. Give each linking transceiver (and the other members of its channel group) a unique receiver number. This is essential to stop endless loops over the link.

2. Decide on the voting type and configure the members of both channel groups accordingly. Choose switched or distributed if possible. If you need to choose central voting, configure the remote channel group with switched or distributed.

3. Enable and configure RF linking, as follows.

   a. In the RF linking form (Configure > RF Interface > RF Linking), select the **Linking transceiver** check box.

   b. In the **Link stream source** box, select the source type that will be assigned to streams that the TB9100 produces from calls it receives on its RF interface.

   - If the TB9100 is the local transceiver, select **Subscriber**.
   - If the TB9100 is the remote transceiver (connected to the remote base station), select **Dispatch**.

   c. If the TB9100 is the local transceiver, enter into the **Link speech impairment** box a value between 0 and 15. This gives the received voice streams a fixed impairment value. Enter a low value if you want the link stream to normally win the vote or be selected in preference to calls from SUs at the main channel group. Enter a high value if you want the link stream to normally lose the vote or not be selected.

   d. If the TB9100 is the remote transceiver, the Link speech impairment box has no effect. The impairment value is not used when selecting dispatch voice streams.

4. Enable RF repeat for each linking transceiver, regardless of the setting of other members of its channel group

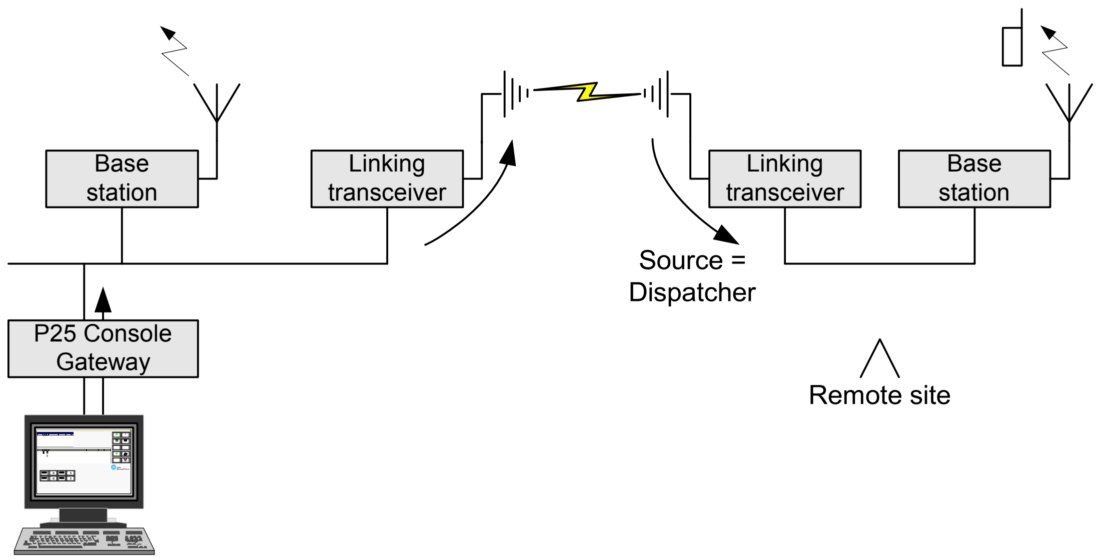
5. Set the preamble length based on the jitter on the linking transceiver's IP link.

6. Decide whether the linking transceivers will use specific NACs or whether they receive any NAC and transmit the NAC in the voice stream, and program them accordingly.

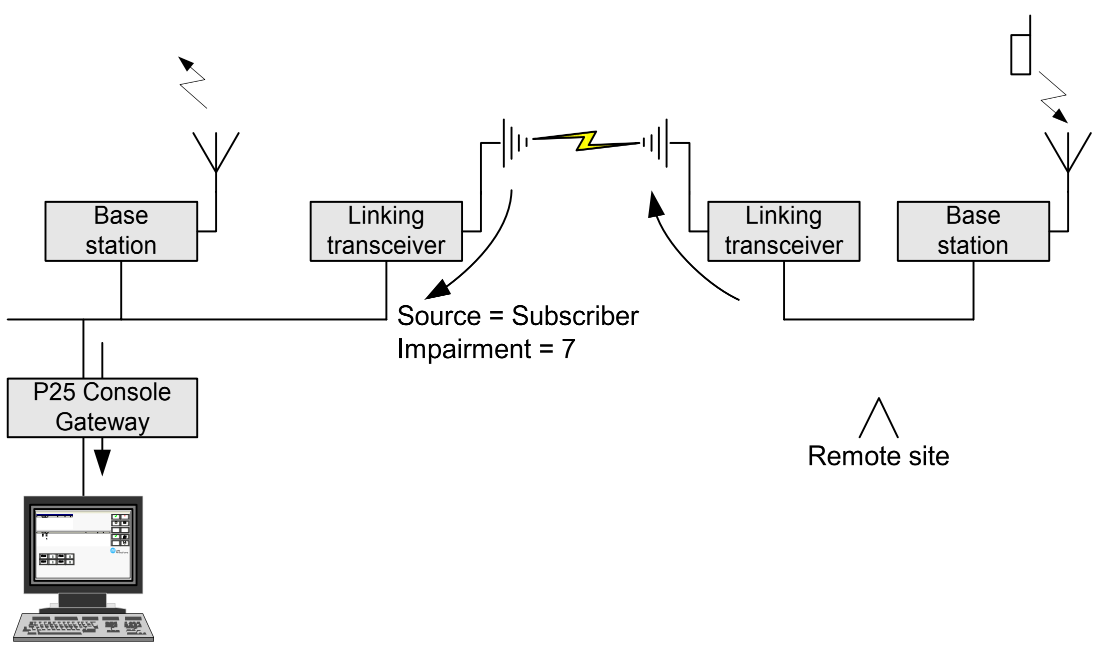
Call Handling

The RF link handles calls as follows (assuming a situation with a single base station at the remote site).

When the dispatcher makes a call, the main channel group's RF interfaces all transmit it. This is because RF interfaces give it a higher priority than any subscriber calls. The local linking transceiver also transmits the call. When the remote transceiver receives the call, it flags it as coming from the source 'Dispatcher.' The remote base station also transmits the call, because its RF interface also gives dispatcher voice streams priority over any subscriber calls. The result is that the local and the remote channel groups transmit the same signal.



When a SU at the remote site makes a call, the remote base station repeats it and the remote transceiver transmits it. The local transceiver flags the call as coming from the source 'subscriber,' gives the call an impairment value and sends it to the main channel group. This call information is provided by the local transceiver's configuration. The local transceiver is configured to flag all calls in this way, because the RF link cannot carry voting and switching information. The main channel group also repeats the call from the remote site and provides it to the dispatcher.



Understanding and Optimizing RF Linking

To understand RF linking and its limitations, it is best to think of the RF link as joining two channel groups (even though one end may be a single TB9100). Voting and selecting occurs at both ends of the RF link.

Because the RF link is not as fully featured as an IP link, it has built-in limitations. The RF link cannot pass information in the voice stream that is used for voting and switching. It can only supply an inbound or an outbound stream to the channel group, not both simultaneously, as an IP-based link can in a duplex channel group.

A combination of product design, configuration choices, and system implementation is needed to optimize RF linking and ensure a satisfactory solution that minimizes the impact of the limitations of the RF link.

The following areas must be addressed.

Voice Streams

The RF link is duplex in the sense that it provides a channel in both directions simultaneously. However, it is an unusual feature of duplex channel groups, that a member may need to provide two voice streams (one inbound and one outbound) to the channel group. The RF link cannot support this feature. It can provide the main channel group with an inbound stream or an outbound stream, but not both at the same time. The product design lets you configure the linking transceivers to flag the streams they receive either as coming from a dispatcher (outbound) or from a subscriber (inbound). The voting and selecting of streams that have crossed the RF link is based on this fixed information.

On the positive side, the RF link can provide the dispatcher with a SU call while the dispatcher is making a call. If this behavior is desired, the channel groups should be configured duplex.

Network Delay

An RF link adds a delay of 60 ms plus the configured preamble duration (usually the minimum of 40 ms). With central voting, this additional delay causes no difficulties if there are no voice streams from the local channel group competing for the vote. If the local channel group does produce one or more voice streams, the voice stream from the remote channel group cannot win the vote. It arrives at the central voter too late for skew compensation and is discarded.

With distributed voting, the additional delay causes no difficulties if the coverage areas of the two channel groups are disjoint. If they are not disjoint, voting between voice streams from the different channel groups may not be satisfactory due to the additional delay. However, distributed voting can still be a good configuration choice if it is rare for the remote base station and a base station in the channel group to be receiving the same call.

Voting

The fact that the RF link cannot carry voting and selecting information limits the capability of the system. The result of voting and selecting can sometimes be unexpected.

If both channel groups have line interfaces and RF interfaces, voting can give one result at one channel group and a different result at the other. RF linking is therefore not suitable for this situation.

If one channel group has a line interface and an RF interface, all streams it sends across the RF link can be flagged as coming from the dispatcher, which means they always are selected in preference to local SU calls. There can be other minor problems as well. For example, if a maintainer uses the control panel microphone at the remote site, that will be selected at the remote channel group but may be selected over SU calls at the main channel group.

An RF link works best with the least sophisticated voting type. The voting type should be the same at both channel groups (except in the case of central voting).

Switched voting assumes that different streams are from different talkers and does not choose continuously between them. In most cases, the winner is the first stream present. The impairment value is only used to choose a stream if two receivers simultaneously provide voice streams.

Distributed voting re-selects the best stream about five times per second. Because the RF link does not carry signal impairment information, the receiving link transceiver adds a fixed value from its configuration. A low impairment value means that the signal from the link is likely to win most votes. A high impairment value means that the signal is likely to lose most votes.

Central voting works least well with RF linking. If the main channel group uses central voting, select switched or distributed voting for the remote channel group.

RF Repeat

Generally, the members of a channel group have RF repeat enabled. They repeat the vote-winning RF signals that they or other channel group members receive. Linking transceivers should repeat RF signals that other members of the channel group received, but not ones that they themselves have received, otherwise they would receive and send the same packets in a never-ending loop.

This problem is solved by product design. When a TB9100 is configured as a linking transceiver, it will not transmit voice streams that are marked with its own receiver number.

Network Address Codes

There are two main ways of using NACs in an RF link.

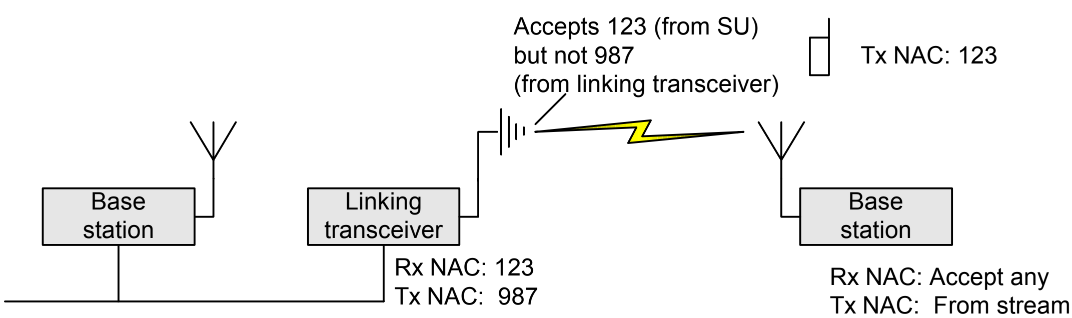
- Use a different NAC for each direction of the RF link. For example, the local transceiver could transmit with the NAC 293 and receive with 283.

The remote transceiver would transmit with 283 and receive with 293. In this scenario, the network cannot carry NACs transparently between the SU and the dispatcher. You must use this method if one end of the RF link is a repeater (see "Using a Repeater as One End of the Link" on page 64).

- Alternatively, allow the linking transceivers to receive any NAC and transmit the NAC that is in the stream. This allows network designers to distinguish groups of users by NAC but it makes the RF link more susceptible to interference from other P25 systems.

Using a Repeater as One End of the Link

When an ordinary TB9100 base station is used as one end of the link, network designers must take a further step for the link to work properly. Not only must the one remaining linking transceiver not re-transmit signals that have come from the link, it must not accept them and pass them to its channel group either, if they were originally transmitted by the linking transceiver. This can be done using NACs. The linking transceiver transmits a different NAC to the one it receives on. When the base station at the other end of the link repeats the linking transceiver's signal, the linking transceiver does not accept the signal back, because it is configured to receive a different NAC.



Using a repeater at one end of the link has a further limitation. The repeater cannot distinguish local calls and calls received over the link. Because calls received over the link are likely to have a high signal strength, they may be given priority over local calls that they do not deserve.

3.5 Firewall

Where a TaitNet P25 network is connected to an organization's intranet or to the public Internet, a firewall is required 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.

The firewall needs to let the following packets through:

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

3.6 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. This access can also be made available to dealers if they are providing ongoing support. A TCP/IP connection is required and this enables Tait to do the following:

- use the CSS to monitor and carry out diagnostic tests on any TB9100 base station on the network.
- use the CSS to remotely upgrade the base station or P25 Console Gateway firmware
- remotely access the base station command line. For this, a secure shell (ssh) session is required.
- remotely access the router command line. For this, the router must be configured for secure shell (ssh) access and have the IP security feature license enabled.

There are two common remote access methods:

- VPN (virtual private network) via the Internet
- Dial up modem/router via the public telephone network

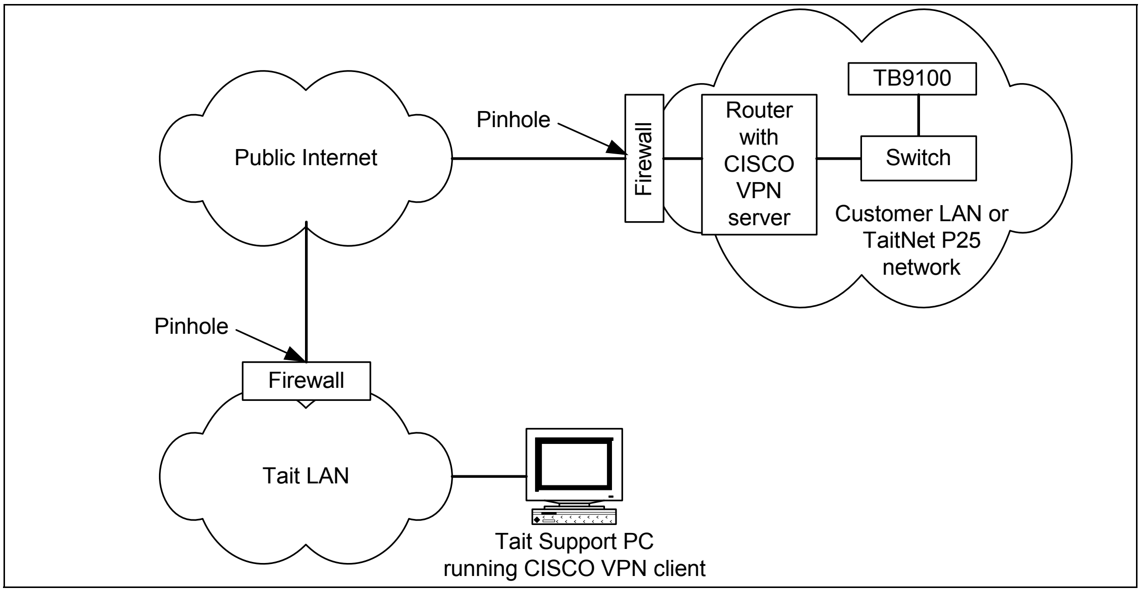
	VPN Access	Dial-Up Access
Setup cost and effort	Moderate	Low: no IP infrastructure required
Operating cost	Low	Can be high
Performance	Broadband speeds (depending on access at customer location)	Limited to modem speeds
Reliability	Good	Varies, can be poor
Security	High	High

Alternatively, Tait may be able to work with the customer's existing remote access solution.

VPN access

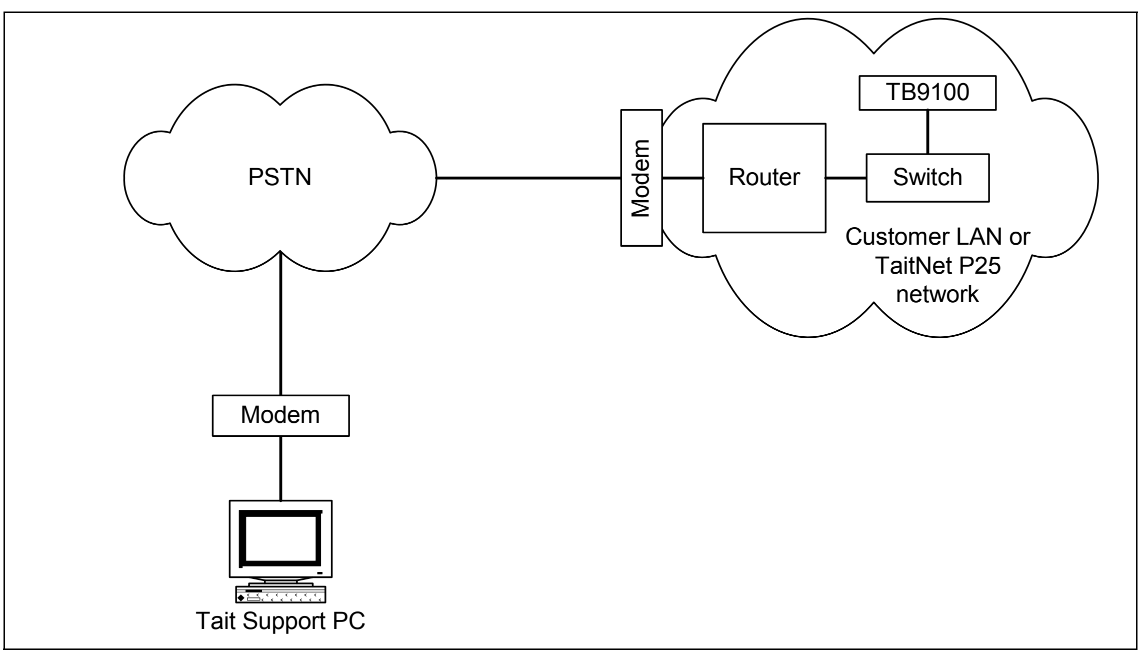
VPN access creates a secure private tunnel through the public Internet. Cisco VPN server software runs on a Cisco router at the customer's network. Those needing to remotely connect to the network use a Cisco VPN client. Communications via this tunnel are encrypted. Unauthorized access is prevented by firewalls at each end of the link. Pinholes in the firewalls allow access to specific IP addresses and pre-defined ports (those used for the VPN transport).

For additional security, customers can unplug the cable connected to the router when remote access is not required.



Dial-Up Access

A modem at the customer site can be dialed from any telephone on the PSTN. Dial-up networking authentication (login/password) is then required to establish an IP connection. Access from a CSS to any base station on the network is protected by the base station password. For additional security, customers can unplug the modem when remote access is not required.



3.7 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. This can be done in various ways. Generally, the first step involves adding Tait P25-capable SUs onto the existing system. The transition can be done channel group by channel group; there is no need to change over the entire system at once. The following tables give an overview of some options.

Option	Step 1	Step 2	Step 3
Transition using P25 dual mode SUs	Add Tait P25 SUs (dual mode) onto the legacy base stations	Turn off legacy base stations. Turn on Tait P25 base stations (dual mode). SUs transmit analog FM, dispatcher selects analog FM using a function tone.	SUs switch to a digital zone. Dispatcher selects digital P25 using a function tone.
Transition using analog FM mode	Add Tait SUs (analog mode) onto the legacy base stations	Turn off legacy base stations. Turn on Tait P25 base stations (analog FM mode)	Change Tait base stations and analog gateway to digital P25 mode. Simultaneously, SU users switch to a digital zone.
Go straight to digital P25 mode	Add Tait SUs (analog mode) onto the legacy base stations	Turn off legacy base stations. Turn on Tait P25 base stations (digital P25 mode). Simultaneously, SU users switch to a digital zone	
Run both infrastructures	Add Tait SUs (analog mode) onto the legacy base stations	Turn on Tait base stations (digital P25 mode). Continue running legacy base stations. The Dispatch console patches calls between the two systems (either always or as needed). Dispatcher transmits on both systems	Turn off legacy base stations once all SUs have switched to a digital zone.

Option	Advantages	Disadvantages
Transition using dual mode	Provides the smoothest and most gradual transition.	Dual mode increases the risk of problems with scanning and voting.
	Because of dual mode, any user whose radio has not been re-programmed or who does not switch to digital mode can still hear conversations.	Dispatch console must be set up to use function tones.
Transition using analog FM mode	Avoids risks associated with dual mode. Dispatch console does not need to use function tones.	Users with radios that have not been re-programmed cannot participate at step 3.
Go straight to digital P25 mode	Lowest risk of performance problems Dispatch console does not need to use function tones.	Difficult to synchronize the transition, especially if a spare channel is not available.
Run both infrastructures	Because of the cross-repeating, any user who does not switch to digital mode can still hear conversations.	Additional frequencies are required for new base stations. These may need to be purchased for the transition period.
		False gating at a legacy base station will override digital P25 calls.
		Patched analog FM calls will appear to digital P25 users to come from the dispatch console

Using Spare Channels

Spare or additional channel frequencies make it possible to carry out a step-wise transition. Set up the spare channel as one of the digital P25 channels, carry out the system acceptance tests on it, and then make the channel operational. When users are ready to begin using this channel, they simply change zone (see Example Plan 1 on page Example Plan 1). In the new zone, the same channel number now specifies the new frequency pair and selects digital P25 mode.

Using Zones

The best way for users to change from one set of channel definitions to another is by using the zone selector (for Tait mobiles, program the left menu key to call up the Zone menu, so that users can easily change zone). Users continue to use the same channel numbers with the same meanings, but select a different zone for each stage of the transition. This is easier than, for example, remembering to add 6 to the channel number to select the digital P25 equivalent of an old analog FM channel. A final step can re-program all zones to be the same. This can be done at any convenient time after the transition is finalized.

Other Tips

- Aim to minimize the amount of mobile and portable re-programming. Choose carefully when to install mobiles: recalling the vehicles for re-programming is to be avoided if possible.
- Make sure you have an agreed rollback strategy. For each stage in the transition, there should be a procedure for reverting to reliable operation if problems emerge.
- Changing channel group operation can be done in two ways: by re-configuring each base station in the channel group using the CSS or by tone remote command from the dispatch console. The CSS method is recommended if the channel is being changed, the tone remote command is recommended if it is only the calling profile that is being changed. (Using tone remote commands to change channel requires setting up Task Manager tasks in each base station in the channel group to propagate the command.)

Example Plan 1

The following example plan is based on the 'Go straight to digital P25 mode' option and proceeds channel by channel. System acceptance tests are carried out on each frequency before it is made available to users. Once a set of users have been assigned to the spare frequency, the old frequency is available for setting up as another new digital channel. The channel numbers continue to mean the same thing for users, but the underlying frequencies change.

1. Add Tait P25 dual-mode radios to the existing fleet. These radios are configured with three zones (three different sets of channel definitions, for selection during the transition period). Users begin operating them on Zone A 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 the spare frequency pair (F3) and some sample SUs.
4. Users of the Ops frequency switch to Zone B and begin using F3 as their digital P25 channel.
5. F1 is now spare. Test and commission the channel group that uses F1.
6. Users switch to Zone C and the SWAT team begins using F1 as their digital P25 channel.
7. F2 is now spare. Test and commission the channel group that uses F2.
8. Users can now begin using the New channel.

This is summarized in the following table.

Frequencies	Step 1	Step 4	Step 6	Step 8
F1	Ops	spare	SWAT	SWAT
F2	SWAT	SWAT	spare	New
F3		Ops	Ops	Ops
SU selection	Zone A	Zone B	Zone C	

(Shaded cells indicate analog FM frequencies, clear cells indicate digital P25.)

For this plan, the SUs need to be programmed as follows:

Channels	Zone A	Zone B	Zone C
1 Ops	F1 analog	F3 digital	F3 digital
2 SWAT	F2 analog	F2 analog	F1 digital
3 New			F2 digital

Example Plan 2

Here is an example plan for the ‘Transition using dual mode’ option.

1. Add P25 dual-mode 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.

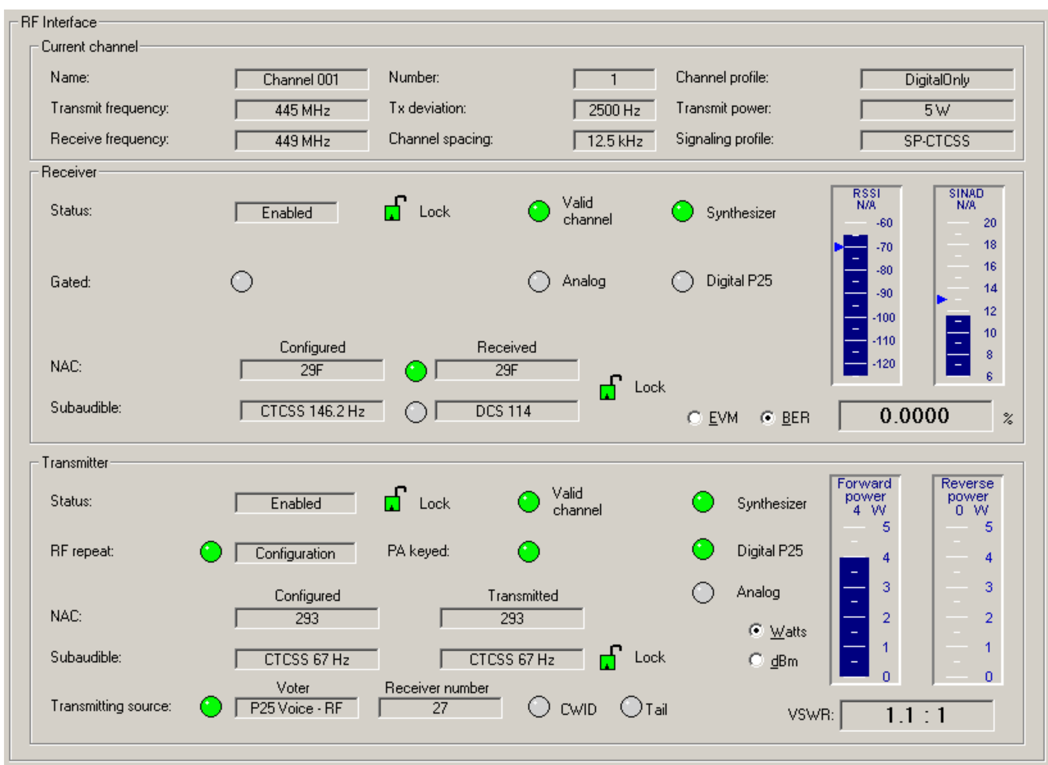
7. Repeat steps 2-6 for each channel group until the entire system is replaced by the TaitNet P25 network.

4 Network Management

The following describes the use of the CSS (in particular for monitoring channel group voting) and the sylog collector. It also provides an overview of SU numbering.

4.1 Using the Customer Service Software

The Customer Service Software (CSS) provides a window into any TB9100 base station or P25 Console Gateway on the TaitNet P25 network. The CSS is a PC-based program that runs under Microsoft Windows. CSS users can remotely monitor, configure, upgrade firmware, and carry out diagnostic tests.



The CSS connects via IP so that it is easy to connect remotely to a TB9100 base station or P25 Console Gateway. You can run several CSS instances on the one PC and connect each one to a different network element.

Monitoring Channel Group Voting

The CSS is also able to monitor channel group voting. You can check the voting configuration, see which voice stream is winning and what other voice streams are present.

To monitor voting

1. Select Monitor > Channel Group > Status.
2. Connect to a channel group member. If voting is centralized, connect to the central voter as this provides more details.
3. Check the configuration items in the lower table to ensure that the channel group members have a consistent configuration.
   - Receiver numbers should be unique.
   - All members should be on the same channel.
   - All should have the same base station mode. None should be in Standby mode. A network element in Standby mode does not participate in voting.
4. Check that all members of the channel group are present. If a member is absent, its link may be down.
5. In the upper table, look for the green cells in the RF and Line In columns. They indicate which member won the RF vote or was selected.

   If you are connected to the central voter, data in other rows indicates other voice streams that lost the vote. If voting is distributed, you can only see the winning stream and the stream being received by the member you are connected to.
6. Check to confirm that the voter is operating as expected (see “Voting criteria” on page 29).

Note If you have only upgraded some channel group members to version 3.05, connect to a version 3.05 member to monitor the channel group. The CSS will display any version 2.2x members that are in the channel group. It will also indicate whether they are operating in simplex or duplex mode, what their RF repeat setting is, and whether they are in Standby mode.

Example Voter Monitoring Displays

The voter monitoring display varies, depending on the type of voting (centralized, distributed, or switched), the type of channel group (simplex or duplex), and the member that the CSS is connected to. The following examples are intended to help you correctly interpret the display.

Centralized voting, duplex channel group

Screens like the following result when the CSS is connected to the central voter. This screen indicates that the Marley's Hill base station is the central voter. The channel group mode is duplex, so that there is voting for the inbound (RF to dispatcher) stream as well as selection of the outbound (dispatcher to RF) stream.

Host Name	Voting	CG mode	RF Repeat	Rcvr Num	RF Rx Type	RF Rx Impmt	RF Rx %	Line In Type	Line In Src	Moni -tor	Skew
Mt Cass	Satellite	Duplex	True	2							20
Kainga	Satellite	Duplex	True	3							20
Mt Cavendish	Satellite	Duplex	True	4							40
PWC	Satellite	Duplex	True	5	P25	10	5				20
Marley's HIll	Central	Duplex	True	1	P25	7	95				0
540 TCL	Satellite	Duplex	True	6				P25	DL		20

Marley's Hill is the central voter. All others are satellites.

Duplex mode means that there are two winning streams, an inbound and an outbound.

Marley's Hill is currently winning the RF vote. In this over, it has won 95% of the time.

PWC is also receiving the call but has only won the vote 5% of the time. Its average impairment is higher. Only the central voter can display losing streams.

A P25 call from the DFSI at receiver no 6 has been selected as the outbound stream.

Skew is only displayed if you are connected to the central voter

Distributed voting duplex channel group

Screens like the following result when the CSS is connected to any member of a channel group with distributed voting. In this case the CSS is connected to the PWC base station. The CSS can only display a vote-losing RF voice stream if the connected base station is receiving the call itself.

Host Name	Voting	CG mode	RF Repeat	Rcvr Num	RF Rx Type	RF Rx Impmt	RF Rx %	Line In Type	Line In Src	Moni -tor	Skew
Mt Cass	Dist	Duplex	True	2							
Kainga	Dist	Duplex	True	3							
Mt Cavendish	Dist	Duplex	True	4							
PWC	Dist	Duplex	True	5	P25	10	5				
Marley's HIll	Dist	Duplex	True	1	P25	7	95				
540 TCL	Dist	Duplex	True	6				P25	DL		

Duplex mode means that there can be two winning streams

Marley's Hill is winning the RF vote 95% of the time. It has the lowest impairment.

PWC is also receiving the call but only wins the vote 5% of the time.

A P25 call from the analog line or DFSI at receiver no 6 has been selected as the outbound stream

Simplex operation

Screens like the following result when the CSS is connected to the central voter in a simplex channel group. There is only one winning stream. A dispatcher call always wins over any subscriber calls, but you can see any losing RF streams.

Host Name	Voting	CG mode	RF Repeat	Rcvr Num	RF Rx Type	RF Rx Impmt	RF Rx %	Line In Type	Line In Src	Moni -tor	Skew
Mt Cass	Satellite	Simplex	True	2							20
Kainga	Satellite	Simplex	True	3							20
Mt Cavendish	Satellite	Simplex	True	4							40
PWC	Satellite	Simplex	True	5	P25	10					20
Marley's HIll	Central	Simplex	True	1	P25	7					0
540 TCL	Satellite	Simplex	True	6				P25	DL		20

Simplex mode means that there can be only one winning stream

The central voter sees two RF streams but neither is selected

A P25 call from the DFSI at receiver no 6 has been selected as the winning stream

Dispatcher selects and disables receivers

Digital dispatch equipment connected via the DFSI can select or disable any receiver in the channel group. Selecting a receiver causes the channel group to declare its signal the vote winner. This also effectively disables all other members. Disabling a receiver excludes it from participation in the RF voting process.

When the CSS is connected to the central voter in a duplex channel group, you see a screen like this:

Host Name	Voting	CG mode	RF Repeat	Rcvr Num	RF Rx Type	RF Rx Impmt	RF Rx %	Line In Type	Line In Src	Moni -tor	Skew
Mt Cass	Satellite	Duplex	True	2							20
Kainga	Satellite	Duplex	True	3							20
Mt Cavendish	Satellite	Duplex	True	4							40
PWC	Satellite	Duplex	True	5	P25	10	100				20
Marley's HIll	Central	Duplex	True	1	P25	7	0				0
540 TCL	Satellite	Duplex	True	6				P25	DL		20

Green means that the dispatcher has selected receiver No 5

Red means that the dispatcher has disabled receiver No 1

Marley's Hill is also receiving the call.

The central voter selected this call as the vote winner, as instructed by the dispatcher

A P25 call from the DFSI at receiver no 6 has been selected as the outbound stream

Dispatcher monitors a base station

Digital dispatch equipment connected via the DFSI can monitor any receiver in the channel group. Asking to monitor a receiver causes it to bypass the normal squelch mechanisms and to provide the voice stream to the dispatcher.

If the CSS is connected to the member with the DFSI interface (540 TCL), it shows whether any monitor (ungated) signal is being sent to the dispatcher (not implement in Version 3.0):

Host Name	Voting	CG mode	RF Repeat	Rcvr Num	RF Rx Type	RF Rx Impmt	RF Rx %	Line In Type	Line In Src	Moni -tor	Skew
Mt Cass	Dist	Duplex	True	2							
Kainga	Dist	Duplex	True	3							
Mt Cavendish	Dist	Duplex	True	4							
PWC	Dist	Duplex	True	5							
Marley's HIll	Dist	Duplex	True	1							
540 TCL	Dist	Duplex	True	6				P25	DL	RF 5	

The selected outbound stream is unaffected by the Monitor command.

The ungated signal from Marley's Hill is being sent to the dispatcher connected to 540 TCL.

4.2 Using a Syslog Collector

Normally, larger TaitNet P25 networks are equipped with a PC running a syslog collector. A syslog collector is a third-party program that collects, filters, and displays alarm and other messages. These messages come (in the industry-standard syslog format) from Tait network elements (base stations and P25 Console Gateways) and also from the TaitNet P25 network's switches and routers. The syslog collector makes it easy to monitor and log the behavior of the whole network from one central point. The syslog collector can also be configured to respond if a network element fails. If it does not receive a regular 'heartbeat' message, it notifies the duty technician.

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). 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).

Configure Kiwi Syslog to display and log the messages you are interested in and to filter out unwanted messages. For example, one display can be set up for call records and another for alarms. You can easily switch between them as needed. In addition, Kiwi Syslog can be set up to monitor base station failure. If Kiwi Syslog does not receive a message with the base station's IP address before a timer expires, it carries out the action you specified (emails the technician, pages the technician, or sends a syslog message). In a similar way, Kiwi Syslog can be set up to notify the duty technician when significant error messages are received.

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.

Call Records

Each channel group member involved in a call generates a set of records for that call. These can be used to identify the calling and called parties, the type of call and its impairment. If the network element is configured with a logging level of Trace, additional call records are produced that show encryption information and handovers in RF voting from one base station to another.

Setting Up Kiwi Syslog to monitor call records

You can configure Kiwi Syslog to display only call records and to log them to their own file. You may also want to provide separate displays for each channel group member.

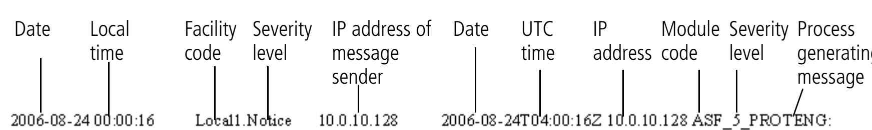
To configure Kiwi Syslog for a call record display

1. Select File > Setup and then select Display from the navigation tree.
2. Modify one of the ten display names to Call Records.
3. Add a rule and rename it Call Records
4. Create a filter that allows only call records to pass, as follows.
   a. Add a filter to the Call Records rule.
   b. Rename the filter Call Records Only.
   c. In the Field drop-down list, select Priority.
   d. Place a tick in each row of the Local1 table. (Call records always have the facility code Local1.)
5. Create an action that displays the filtered call records, as follows.
   a. Add an action to the Call Records rule.
   b. Rename the action Display Call Records.
   c. In the Action drop-down list, select Display.
6. Create an action that logs the filtered call records to a file, as follows.
   a. Add an action to the Call Records rule.
   b. Rename the action Log Alarms.
   c. In the Action drop-down list, select Log to File.
   d. Define the path and the name of the log file.
7. In the main window, select Call Records from the drop-down list to display call records.

Interpreting call records

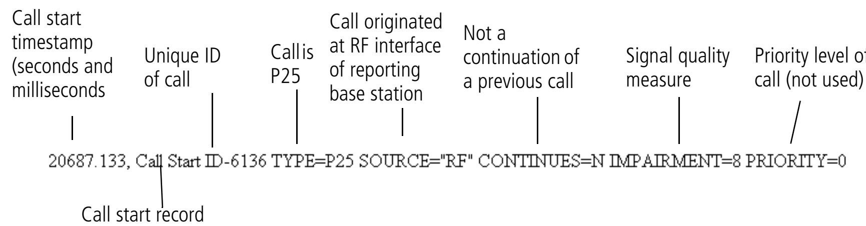
In a normal call, you can expect a call start and a call end message. Other call event messages may occur, for example if there is a handover from one base station to another. As call records are quite long, the following examples divide them into two.

The first fields in each record provide a local date and time (added by the syslog collector) and the IP address of the sender. The date and time is also included later in the message. In this case, they are UTC time.

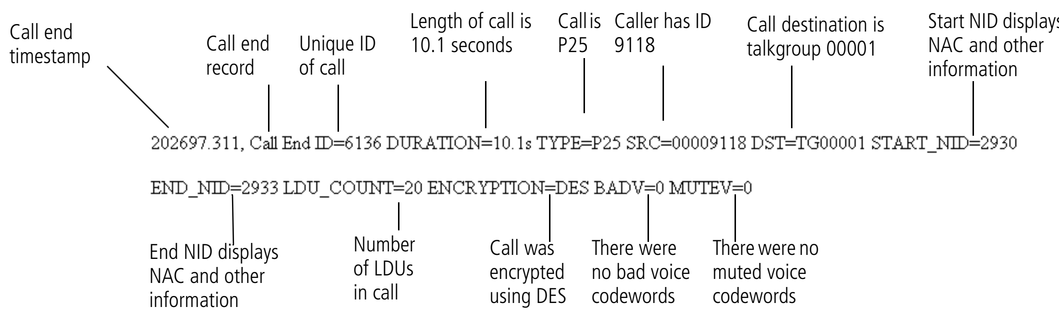


Later fields in the record provide information about the call itself.

The call start record provides the following information.



The call end record provides a lot of additional detail about the call.



Start and end NID

The first three digits of the start and end NID display the NAC of the call (293 in the example above). The final digit has the following meaning.

Digit	Start NID	End NID
7	Call is a TSBK	
0	Call is normal	
3	Call was a terminator only	Call ended normally
5 or A	Late entry	Call faded

BADV and MUTEV

BADV indicates the number of 'bad' voice codewords. These are codewords that needed forward error correction. MUTEV indicates the number of codewords that were muted because they had so many errors that forward error correction could not repair them. Values other than 0 may mean that the calling SU was at the edge of uplink coverage.

Alarms

Each Tait network element (base station or P25 Console Gateway) monitors a large number of operational parameters. When a parameter value crosses its threshold (some parameter thresholds are built-in, others are configurable), the network element generates an alarm message and (if configured to do so) sends it to the syslog collector. Alarm messages are also stored in the network element's own log, which the CSS can display and save to a file. The CSS can also display the current state of all alarms on the TB9100 it is connected to.

Setting up Kiwi Syslog to monitor alarms and other messages

You can configure Kiwi Syslog to display only call records and to log them to their own file. You may also want to provide separate displays for each channel group member.

To configure Kiwi Syslog for an alarm display

1. Modify one of the ten display names to Alarms.
2. Add a rule and rename it Alarms.
3. Create a filter that allows only messages with the facility code Local0 to pass.
4. Create an action that displays the filtered alarms on the Alarms display name.
5. Create an action that logs the alarm messages to a file.
6. In the main window, select Alarms from the drop-down list.

Archiving Log Files

Log files need to be archived, otherwise they keep growing in size until they become unmanageable. Kiwi Syslog can automatically archive log files.

To set up the archiving of log files

1. Select File > Setup.
2. In the navigation tree, right-click Archiving and select Add new archive schedule.
3. Rename and configure the schedule, for example to archive the logs daily into folders with dated folder names.

Example Messages

The following sections describe some important messages and give you some assistance in how to identify and interpret them. Only the MSG part of the records are shown.

Alarm

Standard alarms contain the text FLTLOG, indicating that they are generated by the fault log process. 'ASF' indicates that they originate from the network board. 'REC' indicates that they originate from the digital board. STATE=ACTIVE indicates that the alarm is on. STATE=INACTIVE indicates that the alarm has been cleared.

```
ASF_4_FLTLOG: 00209933.342, 52 - Alarm ASIF_QOS_LOST_PACKETS - STATE=ACTIVE
```

QoS lost packets is the name of this alarm. The CSS manual or online Help provides information about each base station or P25 Console Gateway alarm.

Internal reset

You can search the log for messages that indicate that a reset has occurred. First there is a technical message indicating a failure, followed by a message indicating that the base station or console gateway is starting up.

```
REC_0_UNK: 1156420876.839, REC_0_NOERROR: 00000000.000 0 -   $RCSfile: dblm_common.c,v $#00078
ASF_5_MCNTRL: 1156420894.367, Mode transition completed to STARTING
```

It is important to distinguish internal resets from resets that were requested by a CSS user. CSS resets generate a message like this:

```
ASF_4_SKSRV: 1156421404.863, Base Station reset requested by the CSS
```

Mode transitions

When the base station enters Run or Standby mode, a burst of messages indicate the current operating channel and the transmit and receive frequencies:

```
ASF_5_MCNTRL: 166.206, Mode transition completed to RUN
ASF_5_RXEXIF: 166.435, Digital board status:
ASF_5_RXEXIF: 166.438,    Channel: 3 (Valid)
ASF_5_RXEXIF: 166.441,    Tx frequency: 142020000 Hz
ASF_5_RXEXIF: 166.443,    Rx frequency: 138795000 Hz
ASF_5_RXEXIF: 166.446,    Digital board mode: APCO
```

Task Manager actions

Each time a Task Manager task is carried out, a message at Notice level is generated. You can search for these using the keyword text.

Network board timestamp | Timestamp of TM log | Keyword text | Number of TM task | TM timestamp

```
REC_5_UNK: 302.294, ABS_5_TM_LOG: 00000299.058, 0 - TASK_EXECUTED = 001@000298.806
REC_5_UNK: 302.298, ABS_5_TM_LOG: 00000299.084, 1 - TASK_EXECUTED = 002@000298.806
```

The TM task that has been executed is indicated by a number that is used internally. To find out which task has been executed, count the items in the task list in the CSS, beginning from the top and including any comments, until you reach the task number.

Encryption messages

If encryption or decryption fails, a message at Warning level is sent.

```
ASF_4_PROTENG: 1156346197.505, Cannot decrypt a call with algorithm id 129, key id 3, no license
```

CSS activity

When a CSS user logs on to a Tait network element, a message is sent only if there is a clock synchronization.

ASF_4_SKSRV: 147535.602, BS clock changing NEWTIME=2006-08-24 11:50:30Z TIMESOURCE=10.0.10.2

When the CSS logs off, the following message is sent.

ASF_5_SKSRV: 141136.607, CSS Log Off Request

Notifying 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_5_HRTBT: 00075816.412, 0 - Base Station Alarm is INACTIVE in mode Run

To set up a syslog collector to respond to base station failure

1. For each base station, set up a rule with a filter for any message from the base station's IP address.
2. Set up an additional filter with the Field 'Flags/Counters' and the Filter Type 'Timeout.'
3. Configure the filter to become true if a message is not received in 35 minutes.
4. Set up an action to notify the duty technician (email, pager) if the filter becomes true.

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

4.3 Talkgroup and Individual IDs

SUs and dispatch consoles need to be programmed with individual and group IDs. The P25 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 SU IDs.

If the system has analog dispatch consoles that use MDC1200, these P25 individual and talkgroup numbers need to be mapped to MDC1200 numbers, so that the console can address calls to groups or individuals and display the ANI of calling units. Network elements are provided with a default mapping that is sufficient for networks with up to 1000 SUs.

	P25 Start Address	P25 End Address	MDC1200 Start Address	MDC1200 End Address
Individual	1	999	0001	0999
Group	1	99	9001	9999

In P25, individual and group IDs are distinguishable by the number of bits: individual IDs are 24-bit numbers and group IDs are 16-bit numbers. This means that there can be a group number ‘1’ as well as an individual number ‘1.’ In MDC1200, group and individual IDs share the one number range. In the default mapping shown above, group addresses have 9000 added to the equivalent P25 group address.

If the default mapping is not sufficient for your purposes, you can use the MDC1200 Address Table in the CSS to define additional mappings. If necessary, contact Tait for assistance.

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.

Appendix A — Layered Protocols

Like all modern digital networks, the TaitNet digital network makes use of layered protocols. The protocols specific to the TIA P25 standard or proprietary to Tait make use of the services provided by lower-layer protocols. These lower-layer protocols are the ones used by the Internet. They provide standard solutions to the problems of wide area communication. Internet protocols enable the use of common off-the-shelf equipment such as routers and switches for interconnecting network elements.

The following diagram shows the protocols and the layers they belong to. (The diagram shows five layers, reflecting the reality of Internet protocols. It is a simplification of the classical OSI model, which defines seven layers. Protocol layers are described in most textbooks on networking theory.)

Layer	Protocol	
Application	SSH	Voice service (RTP, RTCP), Syslog, CSS protocol, TCCP, DFSI, NTP
Transport	TCP	UDP
Network	IP (v4)	
Data Link	Ethernet	
Physical		

Physical and data link layers

At the physical and data link layers, network elements make available an Ethernet connection. 10 Base-T cabling suffices for most purposes. As the Internet protocols are hardware-independent, the TaitNet digital network can use a variety of physical links (E1, T1) for switches, hubs, and routers.

Network layer

The Network layer uses IP version 4. This protocol handles data as packets, each of which has a source and a destination IP address. Generally, IP addresses are unicast; they identify a single host. All network elements in a TaitNet digital network have a unicast IP address. However, communications in the TaitNet digital network often use multi-cast IP addresses. Multicast addresses make it possible for many hosts to receive the same set of IP packets. Channel group members send voice streams to the channel group's multi-cast IP address, which all members listen to.

Transport layer

At the Transport layer, there are two Internet protocols: UDP (user datagram protocol) and TCP (transmission control protocol). While most IP-based traffic uses TCP, TaitNet digital network applications use UDP. This protocol is more suited to real-time applications. Moreover, TCP cannot work with multicast IP addresses.

TCP and UDP communications are always to a particular IP address and port. The port number of some applications is fixed, but for others a default is provided and this can be changed using the CSS. If a protocol needs to cross a firewall, a pinhole can be opened for the relevant IP address and port.

Application	Port number	Description
CSS	27,100	Fixed
SSH	22	Fixed
RTP	27260	Configurable
RTCP	27261	Configurable
Syslog	514	Fixed
DFSI control service	7000	Configurable
DFSI voice service	None	Provisioned by FSH on connection
TCCP control service	27258[a]	Configurable
TCCP voice service	27260	Configurable

a. This sets the port number that the gateway network element listens to. When sending, it uses the IP address and port number provided in the message it is responding to.

Application layer

At the Application layer, voice is transported over the network using the RTP protocol. The RTP protocol provides transport for real-time applications such as voice and video. It remedies some of the limitations of UDP by providing detection of lost packets and out-of-order packets.

RTP is a generic type of protocol and allows the definition of different profiles that indicate the format of the payload carried by RTP packets. TaitNet digital networks use proprietary RTP profiles, one for IMBE (digital P25 voice) and another for G.711 (analog FM voice).

RTCP (RTP control protocol) is a companion protocol to RTP. The members of a channel group send RTCP packets to each other. This makes it possible for them to monitor the loss, delay, and jitter characteristics of the network. The CSS can display this information.

Channel group members that are connected to digital dispatch equipment also need to support the DFSI (fixed station interface). This has a control protocol and a voice transport protocol. The voice transport protocol also uses RTP over UDP and is very similar to the protocol used between channel group members.

TCCP (trunking controller control protocol) is used in trunked networks for communications between channel groups and a trunking site controller.

NTP (network time protocol) is used to synchronize the clocks of hosts in a network. It is important in simulcast TaitNet P25 networks because these require highly accurate time-stamping of voice packets.

SSH (secure shell) is used for a secure remote login to network elements.

TFTP is used for remote file transfer.

Appendix B — Syslog Message Format

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.

The syslog protocol defines the following severity levels, in order of decreasing severity:

Level	Name
1	Emergency
2	Alert
3	Critical
4	Error
5	Warning
6	Notice
7	Trace

All TB9100 alarms have a severity level of Warning or higher. The TB9100 may generate a large number of syslog messages at the severity level of Notice.

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	Module Code
TB9100 Base Station (General)	ABS
P25 Console Gateway (General)	AGW
Network board	ASF
Digital board	REC
Power Amplifier	PA
Power Management Unit	PMU

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.

TaitNet P25 Glossary

This glossary contains an alphabetical list of terms and abbreviations related to the TaitNet P25 network, the CSS, the TB9100 base station, and the P25 Console Gateway.

A

administrator A special type of access to CSS functions, used for activities such as changing passwords.

access code A password required to gain access to a set of privileges.

ADC Analog-to-Digital Converter. A device for converting an analog signal to a digital signal that represents the same information.

AES AES (Advanced Encryption Standard) is an encryption algorithm that uses keys of up to 256 bits.

AGC Automatic Gain Control. A device that optimizes signal level.

Algorithm ID The Algorithm ID is an identifier that specifies an encryption algorithm (for example, DES or AES).

analog FM mode A mode of operation in which the RF interface transmits and receives analog FM signal. The network element's channel group interface sends and receives the analog signal as G. 711 speech packets.

analog valid Analog valid is a signal that indicates that the TB9100 base station or P25 Console Gateway is presenting a valid output on the analog line. This output can originate from an analog FM or from a digital P25 call. The M-line carries the analog valid signal.

ANI Automatic Number Identification. A service that provides the receiver of a call with a numerical identifier or alphanumeric label of the caller.

antenna relay A DC-powered device that switches the antenna as needed between the base station's receiver and transmitter. With an antenna relay, a simplex base station only needs one antenna.

APCO The Association of Public Safety Communications Officials in the United States. The APCO Project 25 standards committee defined the P25 digital radio standard. The standard is often referred to as APCO or P25.

ARP ARP (Address Resolution Protocol) is a IP protocol used to map IP network addresses to the hardware addresses used by a data link protocol.

B

Base station A radio receiver and transmitter that is located in a specific place (at a site) that enables a two-way radio to communicate with a dispatcher or over a larger range with other two-way radios. Specifically, Tait TB9100 equipment in a subrack.

Battery protection mode A PMU enters battery protection mode when it has AC power but its DC power is below the configured power shutdown voltage. In battery protection mode, the PMU will shut down to protect the battery if it loses AC power.

BCD BCD (binary coded decimal) is a code in which a string of four binary digits represents a decimal number.

bearer network Telecom equipment that is used to carry user data.

BER Bit Error Rate. A measure of the quality of digital transmission, expressed as a percentage. The BER indicates the proportion of errors to correctly received digits in a received signal.

C

C4FM Compatible Four-level Frequency Modulation. A modulation scheme defined in the P25 CAI standard for 12.5 kHz bandwidth.

CAI Common Air Interface. The over-the-air data formats and protocols defined by the APCO P25 committee.

Calibration Software The TB9100 Calibration Software is a utility for defining the switching ranges of the receiver and the exciter and for flattening the receiver response across its switching range. It can also be used to calibrate TB9100 modules.

call A complete exchange of information between two or more parties. A call requires a receive signal path and a transmit signal path. In trunked systems, a call may be a conversation, made up of a number of overs, but in conventional systems, a call is an over.

calling profile A group of configuration settings that defines the properties of the TB9100 analog line, which can be regarded as equivalent to a SU on the network.

central voting Voting that is centralized at one member of the channel group.

channel

A channel is:
1. A path through which signals can flow.
2. In the RF domain, a frequency pair (or just a single frequency in a simplex system).
3. A set of configuration information that defines the frequency pair and other related settings (a channel configuration). 'Channel' has this meaning in the CSS.

channel coordinator

A software module within the reciter or gateway module that propagates dispatcher channel control commands to the channel group. The channel coordinator also ensures that all channel group members have consistent states so that they work together properly.

channel group

A channel group is a single logical channel consisting of a set of base stations. P25 Console Gateways can also be members. The members of a channel group are linked by an IP network and share a common multicast IP address.

channel module

Channel module is a common term used to refer to reciters and gateway modules. TB9100 base stations have reciters and P25 Console Gateways have gateway modules.

channel profile

A channel profile is a named group of configuration settings that help to define the properties of a channel. Each channel in the channel table must have a channel profile assigned to it.

channel seize

Channel seize is a signal received at the analog line interface, requesting the base station or P25 Console Gateway to accept the signal on the analog line as an input into the channel group. An asserted E-line, LLGT, or LLGT following MDC1200 signaling can function as a channel seize signal.

channel spacing

Channel spacing is the bandwidth that a channel nominally occupies. If a base station has a channel spacing of 12.5 kHz, there must be a separation of at least 12.5 kHz between its operating frequencies and those of any other equipment.

channel table

The channel table is the base station's database of channel configurations.

CKR

The CKR (common key reference) is a number used by the key fill device and by the CSS to indirectly refer to an encryption key without using its Key ID or Algorithm ID.

circuit domain

The part of the base station processing functionality that processes speech signal as a continuous stream of bits – a digital circuit. The opposite of packet domain.

community repeater

Repeater that is shared by several user groups.

CODEC

A device which combines analog-to-digital conversion (coding) and digital-to-analog conversion (decoding).

configuration file A configuration file consists of all the configuration settings needed for a base station or P25 Console Gateway, stored as a file in the configurations folder. Configuration files have the extension *.apc.

connection list A connection list contains the names and IP addresses of base stations and P25 Console Gateways that the CSS can connect to.

control bus The control bus is used for communications between modules in a subrack. It is an I2C bus, a bi-directional two-wire serial bus which is used to connect integrated circuits (ICs). I2C is a multi-master bus, which means that multiple chips can be connected to the same bus, and each one can act as a master by initiating a data transfer.

control panel The control panel is an area at the front of the base station or P25 Console Gateway with buttons, LEDs and other controls that let a maintainer interact with the network element.

conventional network Conventional networks are systems that do not have centralized management of channel access. System operation is entirely controlled by system end users.

CRTP Compressed RTP.

crypto module Module for securely storing encryption keys and for encrypting and decrypting signals.

CSS Customer Service Software. Tait PC-based software for monitoring, configuring, and diagnosing a Tait TB9100 base station or P25 Console Gateway.

CTCSS CTCSS (continuous tone controlled squelch system), also known as PL (private line) is a type of signaling that uses subaudible tones to segregate groups of users.

custom action A custom action is a user-defined Task Manager action that consists of more than one pre-defined action.

custom input A custom input is a user-defined Task Manager input that consists of a set of pre-defined inputs that are combined using Boolean logic.

CWID CWID (**C**ontinuous **W**ave **ID**entification) is a method of automatically identifying the base station using a Morse code. Continuous wave means transmission of a signal with a single frequency that is either on or off, as opposed to a modulated carrier.

D

DAC Digital-to-Analog Converter. A device for converting a digital signal to an analog signal that represents the same information.

DCS DCS (digital coded squelch), also known as DPL (digital private line), is a type of subaudible signaling used for segregating groups of users. DCS codes are identified by a three-digit octal number, which forms part of the continuously repeating codeword. When assigning DCS signaling for a channel, you specify the three-digit code.

de-emphasis De-emphasis is a process in the receiver that restores pre-emphasized audio to its original relative proportions.

DES DES (Data Encryption Standard) is an encryption algorithm selected by the P25 standard.

DDC Digital Down Converter. A device which converts the digitized IF signal of the receiver down to a lower frequency (complex baseband) to suit the DSP.

DFSI The Digital Fixed Station Interface connects digital dispatch equipment with a base station or channel group. It is defined in the Project 25 TIA standard.

digital input value A value that the TB9100 base station computes from the state of a configured number of digital inputs. The digital input value is an input into Task Manager.

digital P25 mode A mode of operation in which the RF interface transmits and receives digital signal as defined by the APCO P25 CAI. The digital line sends and receives IMBE speech packets.

dispatcher A dispatcher is a person who gives official instructions by radio to one or more SU users.

distributed voting Voting for the best RF signal that is carried out separately by each channel group member using the same voting algorithm.

dotted quad A method for writing IPv4 addresses. The form is DDD.DDD.DDD.DDD where DDD is an 8-bit decimal number.

downlink The transmission path from fixed equipment to SUs.

DSP Digital Signal Processor.

dual mode The ability to operate as a transceiver in two different ways: analog FM and P25 digital. Dual mode equipment can be configured to support either mode or to switch between modes from one over to another.

duplex Providing transmission and reception in both directions simultaneously.

duty cycle Duty cycle is used in relation to the PA. It is the proportion of time (expressed as a percentage) during which the PA is transmitting.

E

E & M
A pair of wires used for DC signaling. For example, the signal to set up a call is often sent from the 'M' (mouth) end of a wire to the other 'E' (ear) end by grounding the wire.

EIA
Electronic Industries Alliance. Accredited by the American National Standards Institute (ANSI) and responsible for developing telecommunications and electronics standards in the USA.

encryption
The coding of voice (or data) into unintelligible forms for secure transmission.

EMC
Electromagnetic Compatibility. The ability of equipment to operate in its electromagnetic environment without creating interference with other devices.

ETSI
European Telecommunications Standards Institute. The non-profit organization responsible for producing European telecommunications standards.

F

FCC
Federal Communications Commission. The FCC is an independent United States government agency that regulates interstate and international radio communications.

Feature Code
Code that identifies a software feature license that can be enabled or disabled using the Software Feature Enabler.

Feature Code Sequence Number
Number that indicates how many times a software feature license has been enabled or disabled.

Feature license key
A set of digits purchased from Tait that is required to enable a software feature license.

FEC
Forward Error Correction. A method of encoding data so that the receiving end is able to correct transmission errors.

fill-in receiver
An additional receiver placed within the coverage area of a base station to receive SU transmissions that are too weak to be received by that base station.

FFSK
Fast Frequency Shift Keying. A modem encoding scheme for carrying data on FM radios.

flag
A flag is a programming term for a "yes/no" indicator used to represent the current status of something. The network element has a set of flags that Task Manager can set and clear.

FLASH Electrically block erasable and programmable read-only memory.

FM Frequency Modulation. Often used as an adjective to denote analog radio transmission.

frequency band The range of frequencies that the equipment is capable of operating on.

front panel The cover over the front of the TB9100 base station containing fans for the PA and PMU.

FSH Fixed Station Host.

function code A value that Task Manager can send to the channel group that can serve as an input to Task Manager actions at other channel group members.

G

G. 711 The name of the ITU standard that defines how speech is digitally encoded (64 kbit, A-law or u-law). When the TB9100 base station is in analog FM mode, G. 711 speech is sent and received on the channel group interface.

gating Gating is the process of opening and closing the receiver gate. When a valid signal is received, the receiver gate opens, letting the signal through.

group call A group call is a call that is sent to more than one SUs simultaneously.

H

heartbeat message A message whose purpose is to indicate to the receiver that the sender is operational.

hiccup mode Many power supplies switch off in the event of a short-circuit and try to start again after a short time (usually after a few seconds). This "hiccup"-type of switching off and on is repeated until the problem is eliminated.

HLGT High level guard tone. A tone that announces the beginning of tone remote signaling.

hostname The unique name by which a network element is known on the network.

hub A unit for connecting hosts together. It sends all incoming Ethernet packets to all the other hosts.

hysteresis Hysteresis is the difference between the upper and lower trigger points. For example, the receiver unmutes when the upper trigger point is reached, but will mute again until the level falls to the lower trigger point. An adequate hysteresis prevents the receiver gate from repeatedly muting and unmuting when the level varies around the trigger point.

I

IMBE Improved Multiband Excitation. A voice compression technology patented by Digital Voice Systems, Inc and used in the vocoders of P25 radios.

impairment A measure of signal quality used in channel group voting. Impairment is inversely related to signal quality. The lowest impairment (0) indicates the highest signal quality. The highest impairment (15) indicates the worst signal quality.

inbound Inbound describes the direction of a signal: from a subscriber unit over the air interface to the fixed station.

inhibit A control command that can be sent across the CAI to inhibit a SU. An inhibited SU appears to the user as if it is powered off.

IP Internet Protocol. IP is a protocol for sending data packets between hosts.

isolator An isolator is a passive two-port device which transmits power in one direction, and absorbs power in the other direction. It is used in a PA to prevent damage to the RF circuitry from high reverse power.

K

kernel The kernel is the core executable of an operating system.

key ID The Key ID is the identifier for an encryption key variable.

key fill device A device such as a Motorola KVL3000+ for defining encryption keys and transferring them into P25 equipment.

keytone A signaling tone that accompanies voice on the analog line and is used to key the transmitter. Also referred to as LLGT.

key variable The key variable is a parameter used by the encryption algorithm to encrypt or decrypt a message.

L

LAN Local Area Network

LDU Link Data Unit. Voice calls are sent over the CAI as a series of LDUs.

LED Light Emitting Diode. Also the screen representation of a physical LED.

LLGT Low level guard tone. One of a set of tones used to remotely control base stations.

M

MDC1200 MDC1200 is a proprietary signaling protocol developed by Motorola and used in analog PMR to provide subscriber signaling.

monitor The Monitor function unmutes the receiver, so that the user can hear all traffic on a channel.

multicast group The group of hosts associated with a specific IP multicast address.

multicast IP address An IP address that refers to a group of hosts rather than a single host. These hosts will all accept packets with this IP address.

mute A mute prevents audio from being passed to the radio's speaker.

N

NAC Network Access Code. The 12 most significant bits of the network identifier information that precedes every packet sent on the CAI. The NAC identifies which network the data belongs to, allowing base stations and mobiles to ignore packets belonging to interfering networks.

NAT NAT (network address translation) allows the use of a single IP address for a whole network of computers. A NAT sits between the public Internet and the network it serves, and works by rewriting IP addresses and port numbers in IP headers on the fly so the packets all appear to be coming from (or going to) the single public IP address of the NAT device instead of the actual source or destination.

navigation pane The navigation pane is the left-hand pane of the CSS application window. It displays a hierarchical list of items. When you click an item, the main pane displays the corresponding form.

network element A network element is any device that is network-connected. A TaitNet digital network consists of a number of network elements. The TB9100 base station and the P25 Console Gateway are network elements designed and manufactured by Tait.

normal squelch A type of squelch operation in which the receiver unmutes on any signal with the correct NAC (digital P25) or subaudible signaling (analog FM).

O

octet A set of 8 bits.

outbound Outbound describes the direction of a signal: from a fixed station over the air interface to a SU.

over A single transmission, which begins when a user presses PTT and ends when the user stops pressing.

P

P25 Project 25. A suite of standards and requirements endorsed by the TIA and intended for digital public safety radio communications systems.

P25 Console Gateway A Tait network element that acts as a gateway between an analog dispatch console and a channel group.

PA The PA (power amplifier) is a base station module that boosts the exciter output to the required transmit level.

packet domain The speech processing area that deals with speech data that has been collected up into a packet. IP networks convey packets. The opposite of circuit domain.

PCB Printed Circuit Board

PMU The PMU (power management unit) is a module in the TB9100 base station that provides power to the subrack and monitors power conditions. P25 Console Gateways can also have a PMU.

preamble A well-defined signal that is transmitted at the beginning of digital P25 calls to facilitate downlink voting and to allow the transmit buffer to fill.

pre-emphasis Pre-emphasis is a process in the transmitter that boosts higher audio frequencies to improve the audio quality.

privileges A set of access rights to CSS functions. There are Guest, Maintainer, and Administrator privileges.

program The act of sending a configuration data set from the CSS to the TB9100 base station or P25 Console Gateway.

Project 25 A project set up by APCO (the Association of Public Safety Communications Officials International), together with other US governmental organizations, to develop standards for interoperable digital radios to meet the needs of public safety users.

PSTN Public Switched Telephone Network: The public telephone network.

PTT Push To Talk. The button on a SU that keys the transmitter.

Q

QoS Quality Of Service. A router feature that gives real-time data such as voice calls priority over other data.

R

receiver number A unique number assigned to the members of a channel group and used by the DFSI interface.

reciter The reciter is a module of a TB9100 base station that provides both receiver and exciter functionality.

repeater talkaround Repeater talkaround allows the SU user to bypass repeater operation and so communicate directly with other SUs. While repeater talkaround is active, all transmissions are made on the receive frequency programmed for the channel.

reverse tone burst Reverse tone bursts can be used with CTCSS. When reverse tone bursts are enabled, the phase of the generated tones is reversed for a number of cycles just before transmission ceases. If the receiver is configured for reverse tone burst, it responds by closing its gate.

RISC Reduced instruction set computer. A type of microprocessor that recognizes a relatively limited number of instructions. The reciter's digital board and network board both have RISC microprocessors.

router A router is an internetwork packet switch that switches data packets from an input interface to an output interface. The interfaces can be of different types.

RS-232 A protocol for serial communications between DTE (data terminal equipment) and DCE (data communications equipment).

RSSI RSSI (Received Signal Strength Indicator) is a level that indicates the strength of the received signal.

RTP RTP (Real Time Protocol) is an Internet protocol that supports the real-time transmission of voice and data.

Run mode Run mode is the normal operating mode of the TB9100 base station or P25 Console Gateway.

Rx Receiver.

S

satellite voter A channel group member that has delegated voting activity to a central voter.

SAW filter Surface Acoustic Wave filter. A band pass filter that can be used to filter both RF and IF frequencies. A SAW filter uses the piezoelectric effect to turn the input signal into vibrations that are turned back into electrical signals in the desired frequency range.

selectivity The ability of a radio receiver to select the wanted signal and reject unwanted signals on adjacent channels (expressed as a ratio).

selective squelch A type of squelch operation in which the receiver unmutes only on signals that are explicitly addressed to that receiver. This can be done through a talk group ID or unit ID (digital P25) or through MDC1200 signaling (analog FM).

sensitivity The sensitivity of a radio receiver is the minimum input signal strength required to provide a usable signal.

signaling profile A signaling profile is a named set of configuration items related to signaling that can be applied to any channel. Items include subaudible signaling and transmit timers.

simplex Able to provide transmission and reception only in one direction at a time.

SINAD SINAD (Signal plus Noise and Distortion) is a measure of signal quality. It is the ratio of (signal + noise + distortion) to (noise + distortion). A SINAD of 12 dB corresponds to a signal to noise ratio of 4:1.

site 1. The base station equipment at a particular location. This includes power supplies, transmitters, receivers, network interfaces and controllers. 2. The location of that equipment.

SMR Specialized Mobile Radio. A communications system used by police, ambulances, taxis, trucks and other delivery vehicles.

squelch Squelch is a feature of radio equipment. It ensures that the speaker only unmutes when a valid signal is received. To be valid, it must, for example, have the correct NAC.

SSRC Synchronization source. The SSRC is a large number specified by the trunking controller in its connection message. It uniquely identifies voice streams sent from the master base station.

Standby mode Standby mode is a mode of operation in which active service is suspended so that special operations can be carried out, such as programming in a new configuration or carrying out invasive diagnostic tests.

SU Abbreviation for subscriber unit. This is the term used in the APCO P25 standard documents for a two-way radio (generally a mobile or a portable radio) conforming to the CAI specifications.

subaudible signaling Subaudible signaling is signaling that is at the bottom end of the range of audible frequencies. The TB9100 base station supports CTCSS and DCS subaudible signaling.

subtone A subtone (subaudible signaling tone) is a CTCSS tone or a DCS code.

supplementary service A term used in the P25 standards. It refers to a group of services that is additional to the basic service that a telecommunications network provides. Examples include encryption and SU monitoring.

switching range The switching range is the range of frequencies (about 10 MHz) that the radio equipment is tuned to operate on. This is a subset of the equipment's frequency band.

syslog protocol syslog is a standard protocol used for the transmission of event notification messages across IP networks. TB9100 base stations and P25 Console Gateways can send messages such as alarms to an IP address on the TaitNet P25 network. The base station's logs store messages in the syslog format.

syslog collector A program that can receive, display, and log syslog messages from many devices.

T

TaitNet Brand name for any PMR network designed and manufactured by Tait Electronics Limited.

TaitNet P25 network A set of Tait base stations interconnected by an IP network that can carry voice and data traffic.

TB9100 Base Station A P25-compliant base station consisting of the equipment necessary to receive and transmit on one channel. Generally, this means a reciter, a PA, and a PMU. Often abbreviated to TB9100 or base station.

Task action A task action is the second part of a Task Manager task. It specifies what the network element must do when the first part (the input) becomes true.

Task input — A task input is the first part of a Task Manager task. It specifies what must become true before the network element carries out the second part.

Task Manager — Task Manager is a part of the network element firmware that carries out tasks in response to inputs. These tasks are formulated using the CSS.

TCCP — Trunking Channel Control Protocol. A proprietary protocol operating over IP for the exchange of channel control messages between a TB9100 base station and a trunking site controller.

TCP — Transmission Control Protocol. A complex protocol on top of IP for sending reliable streams of data with flow control.

TELCO — Telephone company.

TIA — Telecommunications Industry Association

toggle — The term toggle is used to describe the switching between two states. If something is on, toggling it turns it off. If it is off, toggling it turns it on.

tone — A tone is a sound wave of a particular frequency.

tone remote function tone — An audio tone used for signaling to a TB9100 base station or P25 Console Gateway on the analog line.

TSBK — A TSBK (trunking signaling block) is an over-the-air message format used in digital P25 mode for setting up trunked calls and for supplementary services such as messaging and status updates.

Tx — Transmitter.

U

uninhibit — A control command that can be sent across the CAI to restore an inhibited SU to normal functioning.

UDP — User Datagram Protocol. A simple protocol on top of IP for sending streams of data.

uplink — The transmission path from SUs to fixed equipment.

UTC — Coordinated Universal Time (word order from French). An international time standard that has replaced Greenwich Mean Time.

V

valid signal A valid signal is a signal that the receiver unmutes to. A signal is valid, for example, when it is strong enough to be decoded and when it has the specified NAC.

vocoder Voice encoder/decoder. A processing element that compresses/decompresses the digital voice signal.

voice stream A digitized voice signal that passes through the main switch.

VoIP Voice over IP. The name for the technology that puts speech signals in packets and then routes them over an IP backbone network.

voting Voting is the systematic sampling of a group of channels for the channel with the greatest signal strength. Voting provides wide-area coverage and ensures that as the user moves throughout the coverage area the strongest channel is always available for a call.

VPN Virtual private network. A private communications network used to communicate confidentially over a non-private network.

VSWR Voltage Standing Wave Ratio (VSWR) is the ratio of the maximum peak voltage anywhere on the transmission line to the minimum value anywhere on the transmission line. A perfectly matched line has a VSWR of 1:1. A high ratio indicates that the antenna subsystem is poorly matched.

W

watchdog A watchdog circuit checks that the system is still responding. If the system does not respond (because the firmware has locked up), the circuit generally resets the system.

Z

zeroize To zeroize one or more encryption keys is to render them useless by overwriting the key data with zeros.