

RTS



APPLICATION NOTES & USE CASES

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ODIN





Figure 1. A typical SFP

The new OMNEO Digital Intercom – ODIN – has the ability to grow through software licensing, and by interlinking multiple frames to create a single intercom matrix. A single ODIN has up to 128 ports. One keypanel uses one port. ODIN is equipped with an Inter-Frame Link, or IFL for short. The IFL uses a bidirectional optical fiber link with a signaling rate of approximately 2 Gbit/s.

To connect the IFL, users will require at least one SFP per unit. SFP stands for Small Formfactor Pluggable and it is the physical interface between the electrical and optical signals. An SFP contains a laser that sends out light on one fiber, and a detector that receives light on the other. Thus, the cable has two strands of optical fiber. A typical SFP is shown in Figure 1. Fibers are available with different physical connectors. This one is called LC.

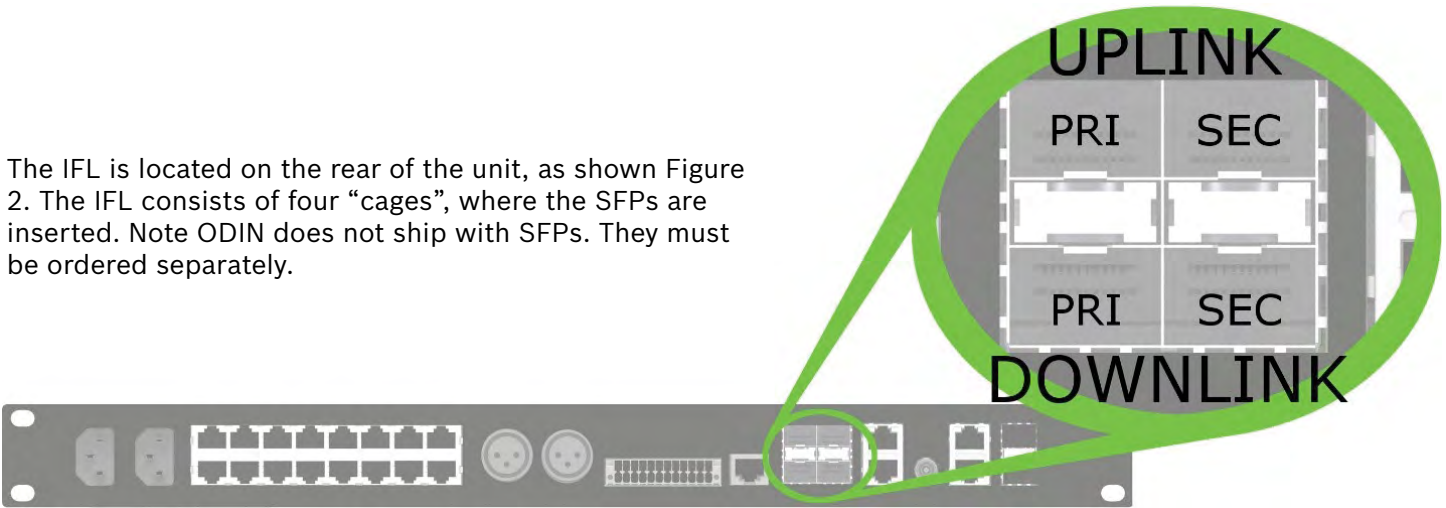


Figure 2. The rear of ODIN has the IFL connectors

In a multi-ODIN configuration, units are connected in a daisy-chain or in a ring. There are two connectors for Uplink and two for Downlink. Each link is duplicated to provide redundancy, but utilizing the redundancy feature is a user choice.

Figure 3 shows the simplest IFL configuration. In the example, four ODINs are interconnected to create a four-frame intercom with up to 512 ports. The Downlink of Frame 1 is connected to the Uplink of Frame 2, the Downlink of Frame 2 to the Uplink of Frame 3, and, finally, the Downlink of 3 to the Uplink of 4. As previously stated, each link is bidirectional, so the Frames are fully interconnected. The IFL transmits audio and some additional information for required for system “housekeeping”. This system is sensitive to a single point failure. If a fiber breaks, the system is split at that point, and the four frames no longer have full interconnectivity.

There are two ways of adding additional redundancy to the system, as shown in Figure 4.

Optical fiber (Inter-Frame Link)

1 Four-frame ODIN system, simple IFL interconnectivity

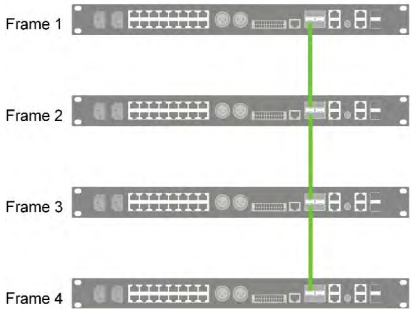


Figure 3. IFL configuration 1

2A Four-frame ODIN system, single IFL ring



2B Four-frame ODIN system, duplicated IFL interconnectivity

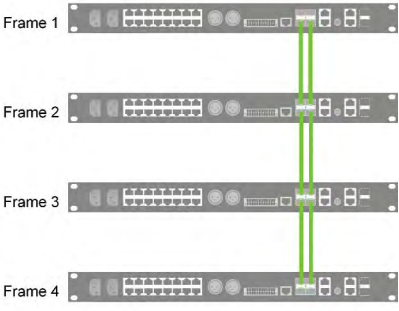


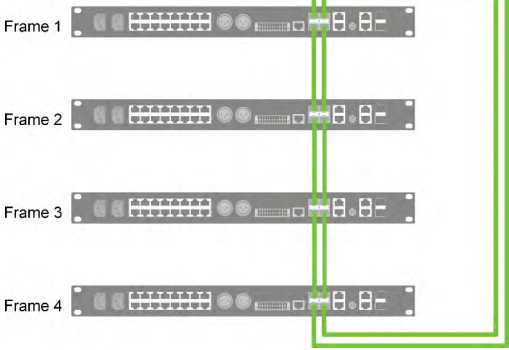
Figure 4. IFL configurations 2A and 2B

In configuration 2A, Frame 1 and Frame 4 are interconnected. The Downlink of Frame 4 goes to the Uplink of Frame 1. This ring is resilient to a single fiber failure. Two additional SFPs are required.

In configuration 2B, each fiber link is duplicated using the Secondary. This requires six additional SFPs. This structure is resilient to multiple fiber failures, as long as two failures do not occur between the same two frames.

Configurations 2A and 2B are not equivalent from a system reliability perspective. Suppose Frame 2 goes off line. Maybe it was in a different rack, and the breaker in that rack tripped. In configuration 2A, you are left with three of the four frames fully interworking. In configuration 2B, you will be left with two islands: there is no audio between Frame 1 and Frames 3 & 4.

3 Four-frame ODIN system, double-ring IFL interconnectivity



Configurations 2A and 2B can be combined, as shown in Figure 5. This double-ring IFL requires a total of 16 SFPs but offers a very high level of resilience against fiber failure.

The level of redundancy should be determined based on the criticality of the application. With ODIN, you have a choice. ODIN also offers a unique degree of scalability, starting with a 16 port single ODIN while allowing growth up to eight ODIN frames with 128 ports each, for a total of 1024 ports.

Figure 5. IFL configuration 3

INTRODUCTION

The OMNEO Digital Intercom from RTS – ODIN – comes with connectors for OMNEO, AIO, and two-wire, also known as wired partyline. A single unit can handle a total of 128 ports. The rear of the unit is shown in Figure 1.

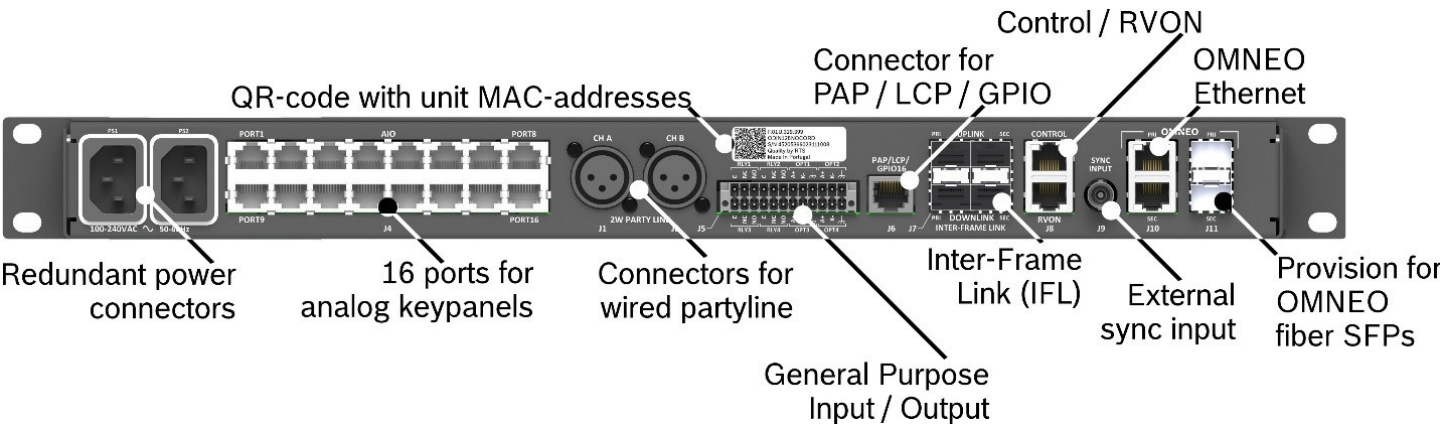


Figure 1. Rear of ODIN showing all the connectors

USING ALL THE CONNECTORS FOR ANALOG AND WIRED PARTYLINE

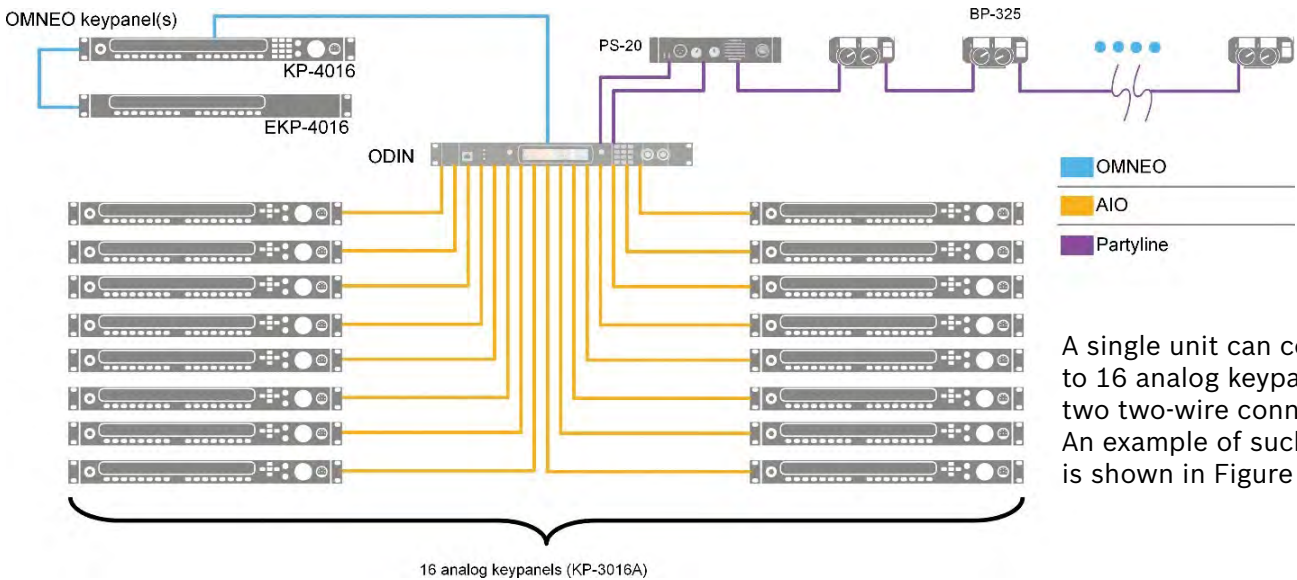


Figure 2. An ODIN with 16 analog keypanels and two two-wire connections

The BP-325 is a binaural beltpack, meaning it uses two channels. That is why it uses both two-wire ports on the rear of ODIN. In some cases, it is necessary to add one or all of the following:

- A few additional analog keypanels
- Many additional analog keypanels
- Additional analog audio sources
- Additional wired partyline connections

ADDING A FEW ADDITIONAL KEYPANELS

The easiest way to do this is by using the OEI-2, the OMNEO External Interface. It converts between the AIO (analog) and OMNEO (digital) interfaces. Both the audio and the digital keypanel data are converted. A single OEI-2 can handle two analog keypanels. Figure 3 shows an example where an additional four analog keypanels are connected using OEI-2.

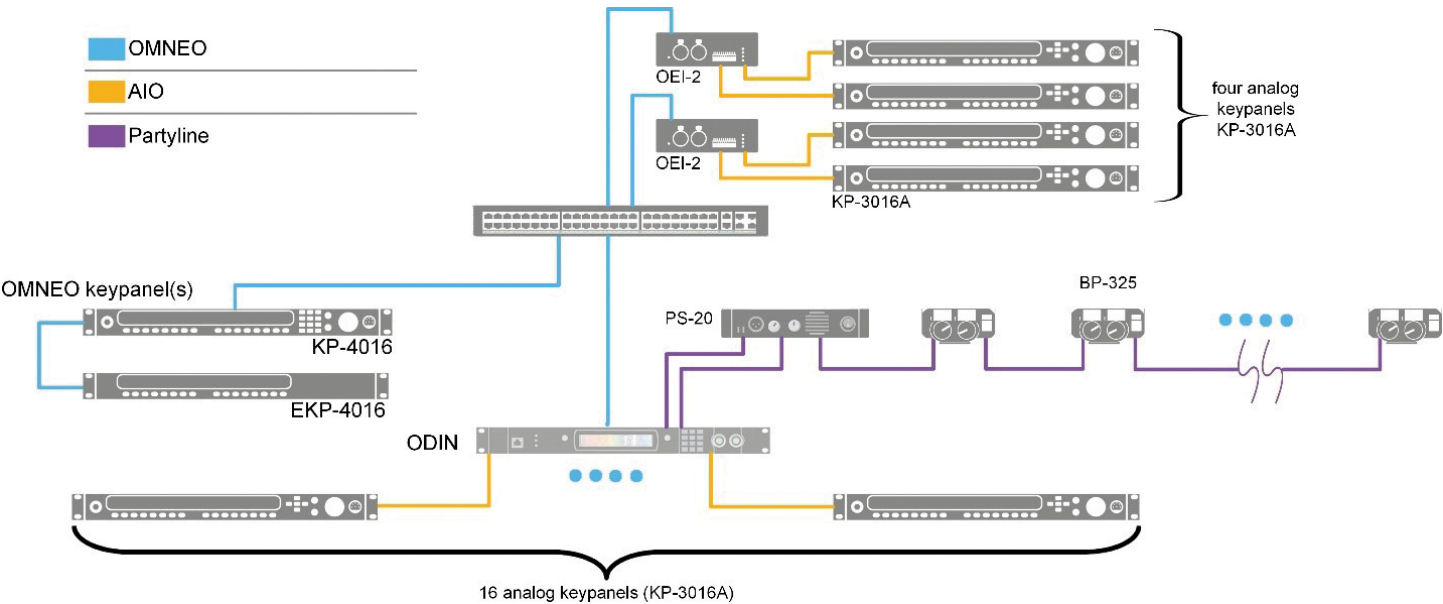


Figure 3. Example with OEI-2 box for adding analog keypanels

ADDING SEVERAL ADDITIONAL KEYPANELS

If you need to connect more than an additional 12 analog keypanels, it makes sense to purchase another 16-channel ODIN and connect the two matrices using the Inter-Frame Link (IFL). That is shown in Figure 4. To learn more about the IFL, read “Application Note: Interconnecting ODIN frames”.

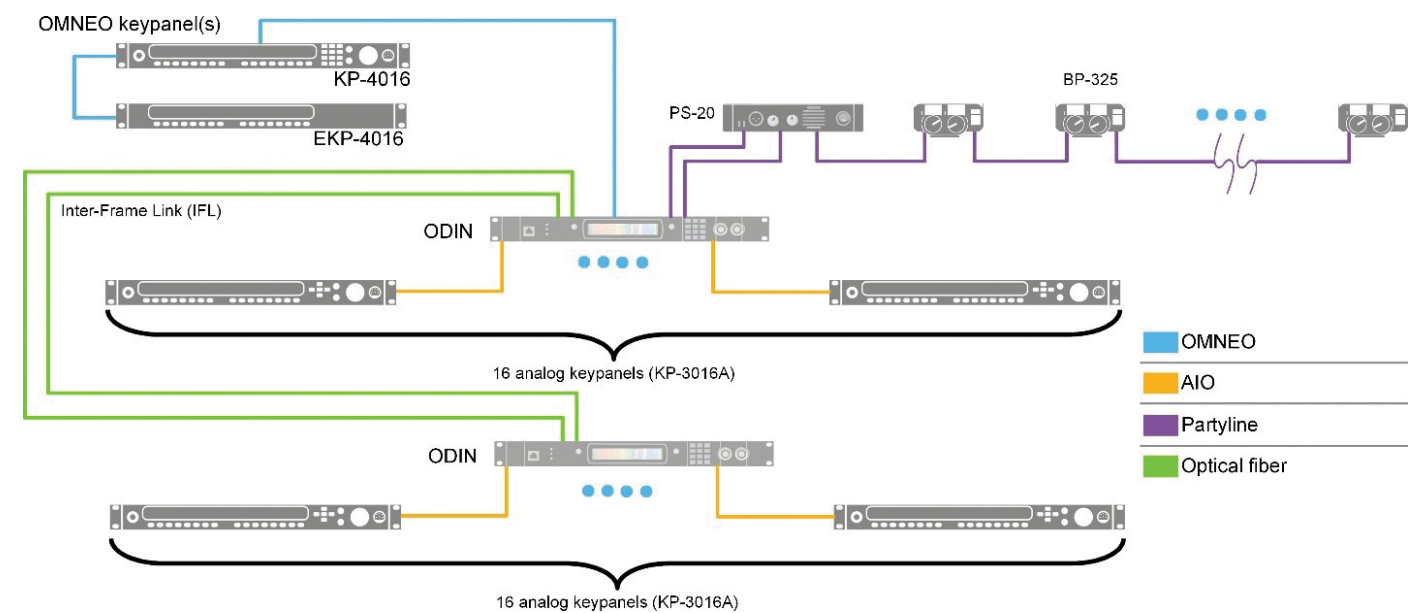


Figure 4. An expanded system using two ODINs

ADDING ANALOG AUDIO AND ADDITIONAL WIRED PARTYLINE

In some cases, it is convenient to bring external audio sources into the matrix. There is no keypanel data to worry about, so an analog to Dante converter can be used for this purpose. For the additional partyline connections, another external converter is used. It converts partyline into Dante (which is compatible with OMNEO). Figure 5 shows this application.

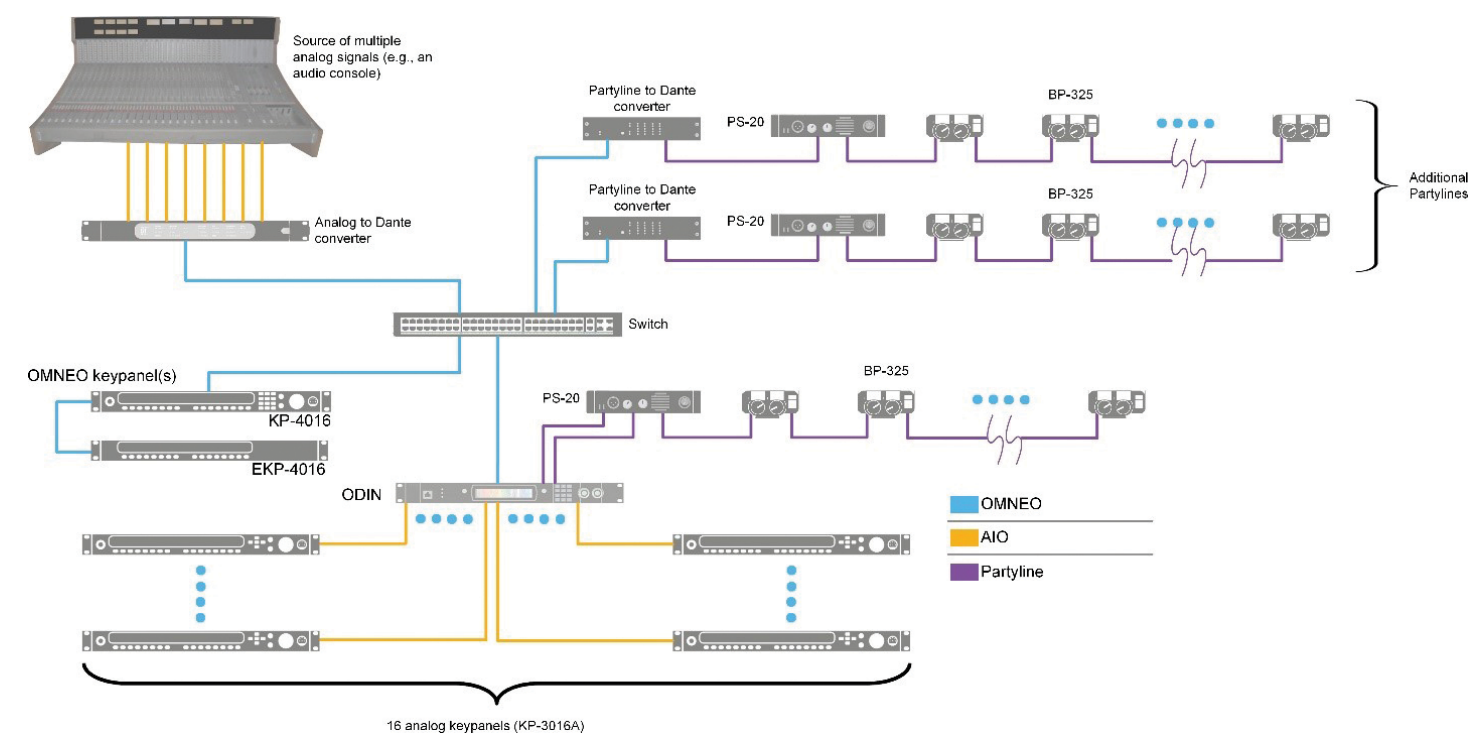


Figure 5. An ODIN system with several external analog audio sources

Converters are available from multiple third-party suppliers. Analog-to-Dante converters are available from Focusrite, Solid Stage Logic, Atterotech and others. The example uses a Focusrite RedNet A16R. The partyline to Dante converter is a Studio Technologies D45R, which works with the RTS two-wire signal format with one or two channels. For information about these products, please refer to the website of the respective manufacturers. RTS has two products that convert two-wire to AIO: the DSI-2008 and the SSA-324.

CONCLUDING REMARKS

ODIN has connectors for a multitude of external device types. Combined with the ability to interconnect multiple matrices (using the IFL), the variations are virtually limitless.

WHAT IS RVON?

RTS Voice Over Network, or RVON for short, is a Voice Over IP (VoIP) solution for RTS products. RVON allows keypanel audio and data to be carried over long distances, using standard Ethernet. There is no distance limitation for RVON: it can be transmitted over the public internet from one end of the world to the other. To achieve this, the audio is compressed, to reduce the required bitrate. Three levels of compression are available. The compression algorithms are known as codecs. You may read more about these in the Application Note called “RTS Voice Over Network – RVON”.

RVON FOR KP-SERIES

RVON firmware is available free of charge for all the RTS KP-Series keypanels. It may be downloaded from the RTS web page. The only keypanel that does not support RVON is the KP-3016A, because it is analog only. We will look at two use cases for RVON.

RVON AS A LONG-DISTANCE VOIP SOLUTION

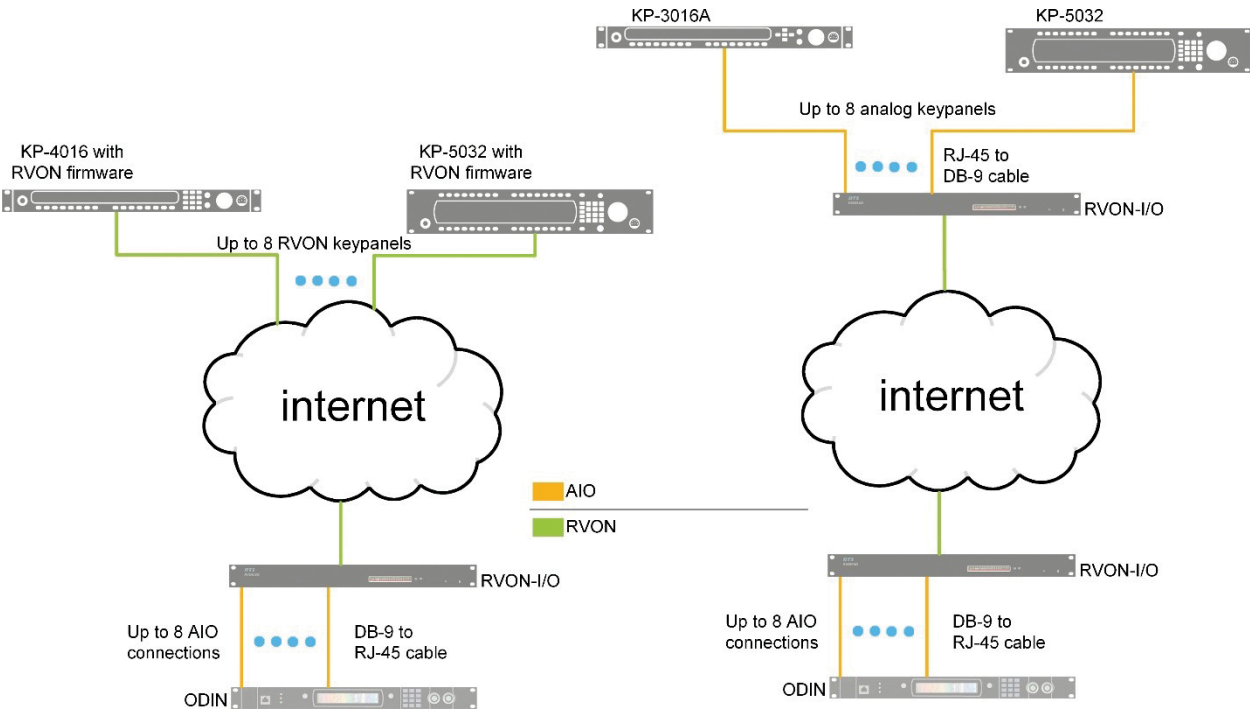


Figure 1. RVON solutions for ODIN

Two use cases are shown in Figure 1: ODIN communicating with RVON-enabled keypanels (left) and ODIN communicating with analog keypanels (right). Both cases require the RVON-I/O, which translates between AIO and RVON.

RVON-ENABLED KEYPANELS

RVON-enabled keypanels may be connected directly to the Ethernet connection (internet in this case). A special cable is required to connect the matrix (ODIN) to the RVON-I/O box. The required cable has an RJ-45 connector on the matrix side, and a male DB-9 connector on the other. The way pins are numbered is shown in Figure 2 and Figure 3.

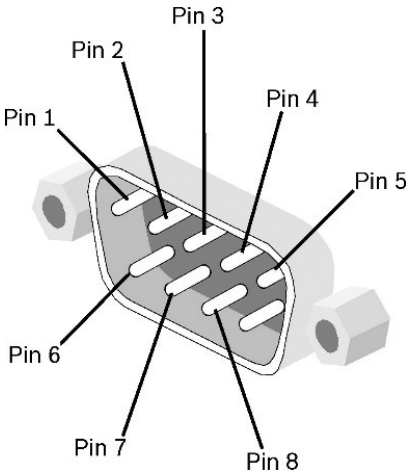


Figure 2. DB-9 cable pinout

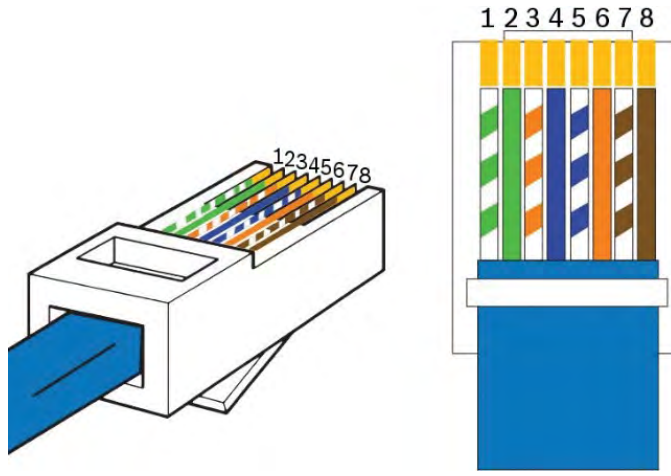


Figure 3. RJ-45 cable pinout

Referencing the numbering in the two figures above, the required connections are shown in Figure 4.

RJ-45 to Matrix		DB-9 male to RVON-I/O
1	NC	
2	Data--	2
3	Audio-from-MX+	4
4	Audio-to-MX+	8
5	Audio-to-MX-	7
6	Audio-from-MX-	5
7	Data+	1
8	NC	

Figure 4. Connections for the cable between matrix and RVON-I/O box

ANALOG KEYPANELS

Keypanels that do not have RVON require an RVON-I/O box as a translator. Up to eight analog keypanels may be connected, as shown on the right hand side of Figure 1. A special DB-9 to RJ-45 cable must be used. Referencing the numbering in Figure 2 and Figure 3, the required connections are shown in Figure 5.

RJ-45 to keypanel		DB-9 male to RVON-I/O
1	NC	
2	Data--	2
3	Audio-from-MX+	8
4	Audio-to-MX+	4
5	Audio-to-MX-	5
6	Audio-from-MX-	7
7	Data+	1
8	NC	

Figure 5. Connections for cable between RVON-I/O box and analog keypanels

INTRODUCTION

ODIN is a fully OMNEO-capable 1RU matrix with the ability to work with ROAMEO, a DECT-based wireless communication system. In this Note we will look at a couple of ways ODIN can support ROAMEO in truck applications.

SCENARIO

In Television production, the term Outside Broadcasting (OB) is used for sports events, music concerts etc not occurring in a studio. OB-vans contain all the equipment required for producing these events and sending them back (via satellite) to the network for distribution. Among the many different types of equipment installed in OB-vans, an intercom system including wireless capability is mandatory. For larger productions, multiple OB-vans are used, and the ability to interconnect the intercom systems becomes important.

BASIC CONNECTION

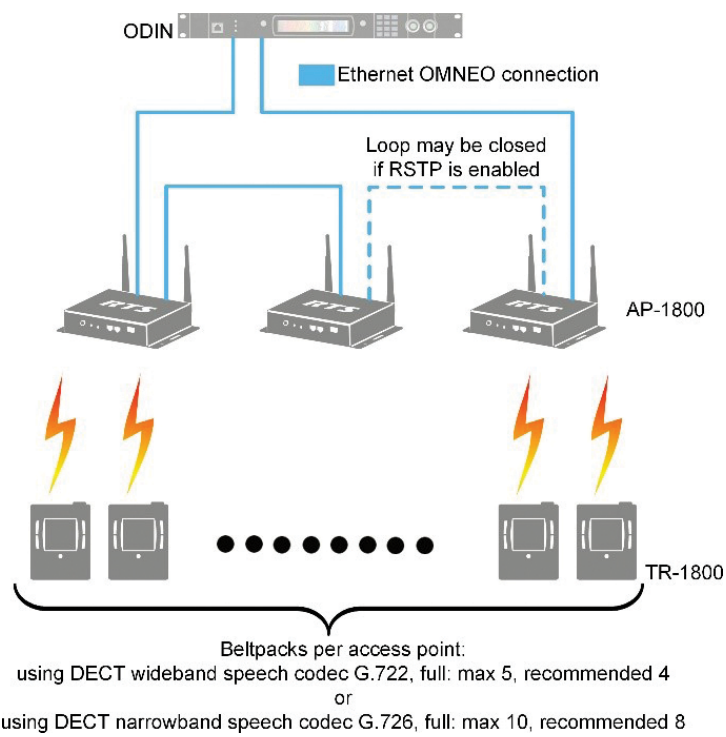
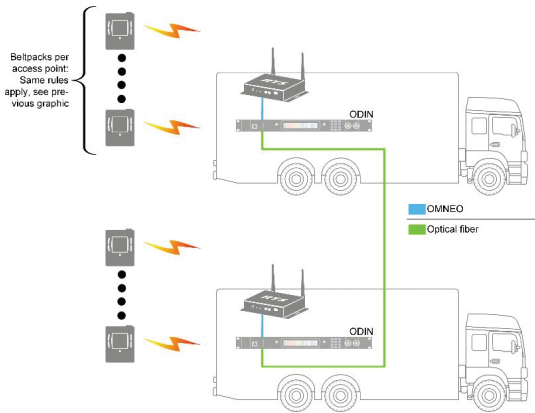


Figure 1. Connecting ROAMEO to ODIN: basic connection

OPTION 1: USING THE ODIN INTER-FRAME LINK (IFL)

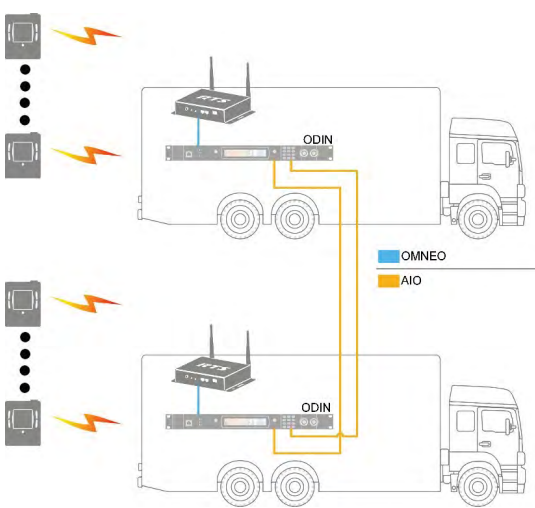
In the following example we imagine two trucks, each with its own intercom including ODIN with one single Access Point of ROAMEO and a number of Beltpacks. The Inter-Frame Link (IFL) may be used to interconnect the two systems. In so doing, they become one single intercom. See Figure 2.



IFL uses optical fiber. Each ODIN must be equipped with at least one SFP, which is the physical device that connects to the fiber. SFPs must be ordered separately, because they are specific to the type of optical fiber used. Ways of interconnecting two or more ODINs using optical fiber are described in the App Note titled “Interconnecting ODIN frames”.

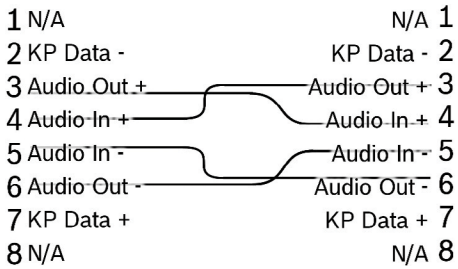
Figure 2. Interconnecting the trucks with IFL

OPTION 2: USING THE AIO CONNECTORS



This option uses two or several analog connectors. ODIN has 16 RJ-45 style connectors for analog signaling, intended for analog keypanels. Of the available 8 pins on an RJ-45 connector, 6 are used. Both keypanel data and analog audio are transmitted using balanced signaling, also known as differential signals, which is more resilient to induced noise. Two wires are used for data, two for analog audio in one direction, and two in the other. When AIO is used for audio signals between two matrices, keypanel data is not transmitted, so only four wires are used. A special cross-over cable must be used, where pins 4 and 5 on each side connect to pins 3 and 6 of the other, see Figure 4.

Figure 3. Interconnecting the trucks with AIO



This interconnect cable can easily be fashioned with an RJ-45 crimping tool to allow the necessary audio cross-over function for the two matrices to talk.

Note the number of AIO-cables used in Figure 3 is not locked to two. The number of cables to be used is an intercom design decision, and is determined based on the total number of required individual (point-to-point) and group (partyline) conversations required between the two intercoms.

Figure 4. Four-wire interconnect cable used in Figure 3

CONCLUDING REMARKS

The solution that uses IFL is more easily scalable to multiple trucks than the one that uses AIO. For multiple trucks, the AIO-based solution requires a fully meshed structure (e.g., each matrix must have direct AIO lines to each of the other trucks to which it wants to talk). With IFL, configuring a multi-ODIN system with ROAMEO becomes a matter of intercom programming only: the intercom behaves like a single matrix, with full access to every beltpack.

RTS



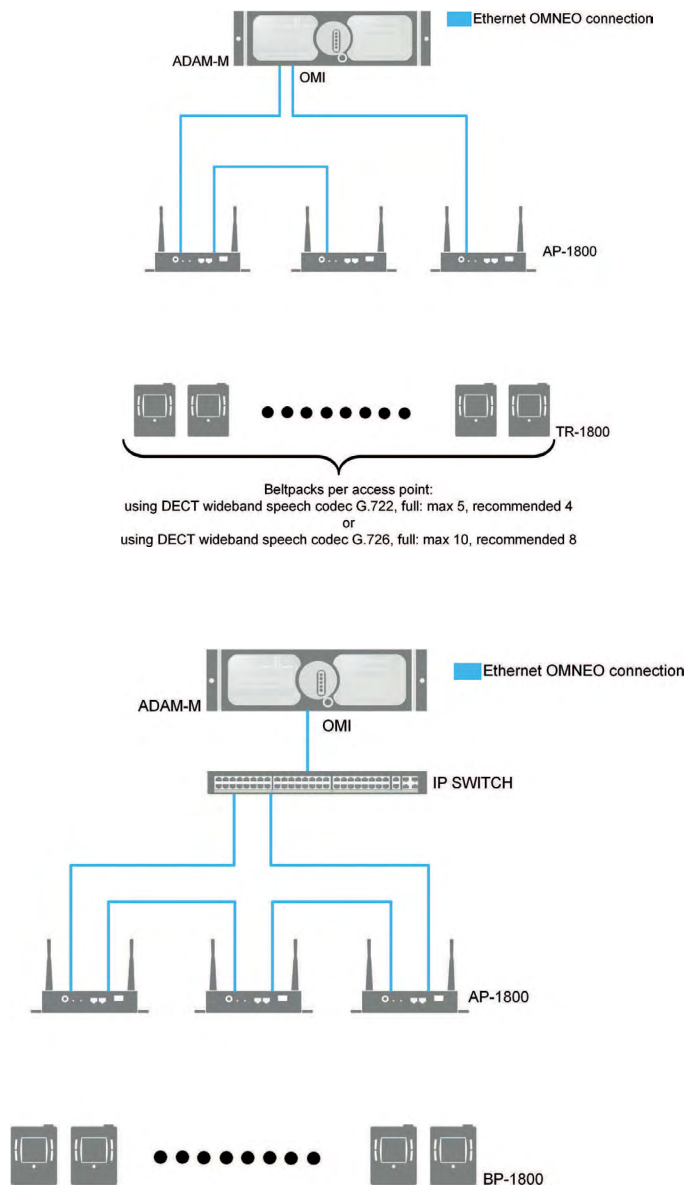
ROAMEO

PROBLEM/SOLUTION

Many users require the features of a new integrated wireless intercom system, but do not want to replace existing equipment. The RTS ROAMEO system is designed to enhance existing investments by making it possible to connect a digital wireless intercom system to the existing matrix. The access points are connected to the ADAM or ADAM-M matrices through the use of an OMNEO Matrix Interface card (OMI). OMNEO is the Dante-compatible solution for high quality, low-latency audio over IP. The OMI card can also be used for both wired keypanels and wireless beltpack connections to the matrix.

PRACTICAL EXAMPLES

Below are three use cases, each of which illustrates a different mechanism for connecting the access points to the matrix. The first is a simple use case that shows ad-hoc wiring. The second uses a ring topology to create basic redundancy. The third uses a sophisticated architecture for a highly redundant network.

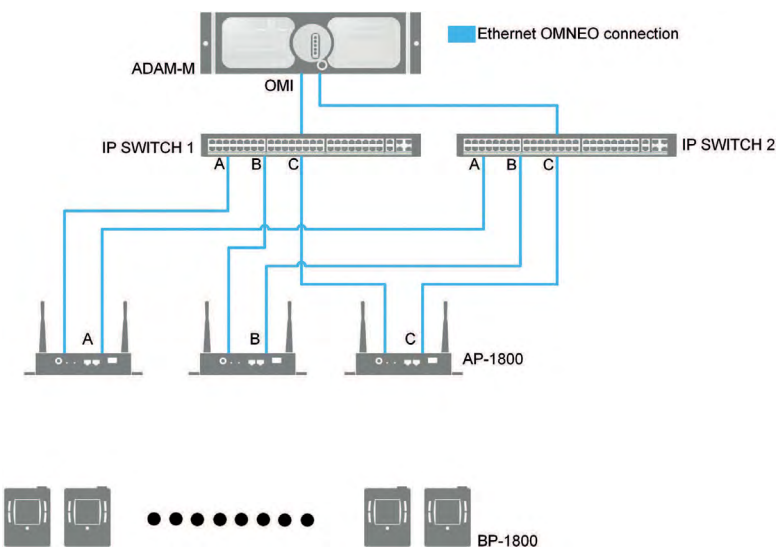


1. Ad-hoc wiring:

An access point can be connected directly to the matrix or to another connected access point, as shown in this example. Each access point has two RJ-45 ports, internally connected through a switch. This implies a ROAMEO system can be easily expanded by adding a new access point and, in most cases, simply connecting it to the nearest existing access point. A chain like this can have up to seven access points. Multiple chains can be created by using an Ethernet switch (not shown in this example).

2. Basic redundancy:

Unlike standard Dante, OMNEO supports a feature called RSTP (Rapid Spanning Tree Protocol). In the example shown here, a level of basic redundancy is achieved by connecting access points in a ring topology. Each access point has two paths for the signal to get back to the switch. If one path fails, RSTP will automatically switch the signal to the alternate path, with a minimum of signal interruption. If the cables break in two locations, this topology can create an isolated "island" of one or several access points that no longer have a path back to the switch. The switch itself is also a single point of failure.



3. Advanced redundancy:

The architecture shown in this example has duplicated switches, both operating with RSTP. Each access point is connected to both of the switches, providing full redundancy on a per access point basis. A single cable failure will not affect the network at all. A double cable failure will potentially isolate one access point, but not more. A very high level of redundancy is provided in this solution, at the cost of an additional switch and more wiring. ROAMEO offers a choice of possible topologies to strike the required balance between redundancy and cost efficiency.

PRODUCT VIEW



AP-1800 uses local power. A PoE adapter kit is available separately. With a PoE adapter, it is possible to drive one AP-1800 from a PoE-enabled Ethernet switch.

INTRODUCTION

ROAMEO is a wireless matrix system from RTS. In this Application Note, we will look at a way of connecting the ROAMEO Access Points using Power over Ethernet, PoE.

WHAT IS PoE?

Power over Ethernet is a way of simplifying wiring for devices that connect to Ethernet. With PoE, there is no need for a separate power supply. There is an upper limit on the amount of power that can be supplied using PoE, and the limit is dependent on the type of PoE switch used. In general, PoE is intended for devices that require little power, up to about 13 Watts. (There is another standard called PoE Plus, which allows up to 25 Watts of power, but we will not get into that here.) With PoE, the data signal is superimposed on the power. On the receiving end, filters are used to pick out the data from the power. When a PoE device is turned on, it communicates with the PoE switch to “negotiate” the right electrical parameters. For this reason, PoE devices can never be connected in cascade.

THE ROAMEO AP

The ROAMEO Access Point, AP-1800, is designed to be tolerant to PoE, but it is not designed to use PoE directly. Expressed differently, it is PoE tolerant but not PoE enabled. RTS does supply a PoE adapter kit, to allow the AP-1800 to work with a PoE enabled switch. The PoE kit is shown in Figure 1.

The kit can be ordered as any other part. The part number is F.01U.335.167 (POE ADAPTER 12V) and it contains the following parts:

- The PoE adapter
- A barrel adapter
- A short DC-cable
- Two Ethernet-cables



Figure 1. The PoE kit for AP-1800

SETTING THE VOLTAGE ON THE PoE ADAPTER

The ROAMEO Access Point is designed to be powered by 12 VDC. Set the Voltage selector on the PoE adapter to 12V. If the PoE adapter is set to a lower Voltage, e.g., 5 or 9 V, nothing will break, but the access point may not work correctly, even though it may seem to power up. See switch location in Figure 2.

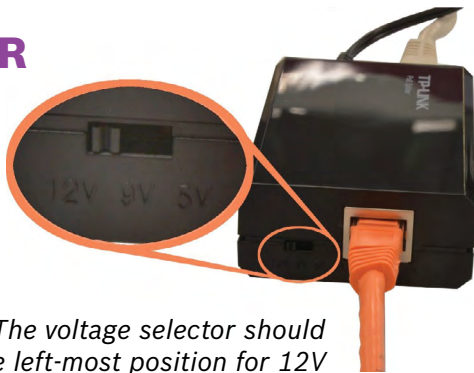


Figure 2. The voltage selector should be in the left-most position for 12V

HOW TO CONNECT THE PoE ADAPTER

Once the voltage has been set to 12 V as described in the previous section, the cable from the PoE switch is connected to the rear of the PoE adapter, and the signal output and DC connectors are connected to the AP-1800. Note you must use the barrel adapter to make a reliable electrical connection between the cable and the AP-1800. Without the barrel connector, it will not work.

We will now look at some ways in which the PoE-enabled Access Point can be connected to the system. To symbolize a PoE-enabled Access Point, we will use the symbol shown in Figure 4.

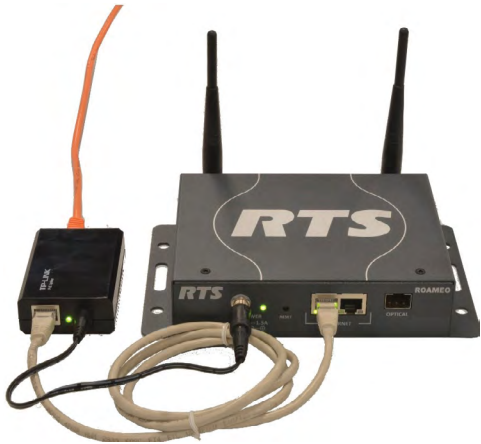


Figure 3. How to connect the PoE adapter

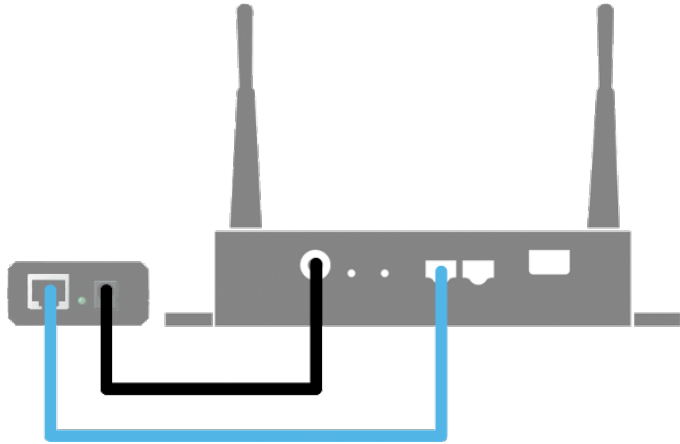


Figure 4. This is the symbol for a PoE-enabled AP-1800

SYSTEM DIAGRAMS

In the first example, we will show how a single AP-1800 may be connected. Note the PoE-enabled AP-1800 on the right is connected directly to the switch. The other two APs are running on local power (meaning they are connected to power using the supplied AC to DC converter). The first AP is connected to PoE, but is not utilizing that power.

In the second example, another AP is connected to the PoE-enabled AP. The second AP must have local power. The OMNEO signal coming out of the second Ethernet jack on the PoE-enabled AP no longer has power.

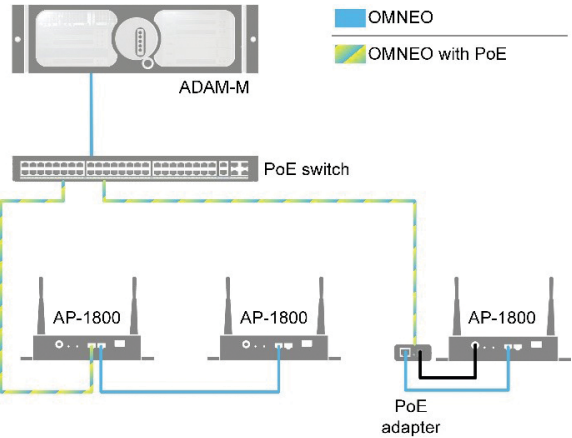


Figure 5. Connecting a single PoE-enabled AP-1800

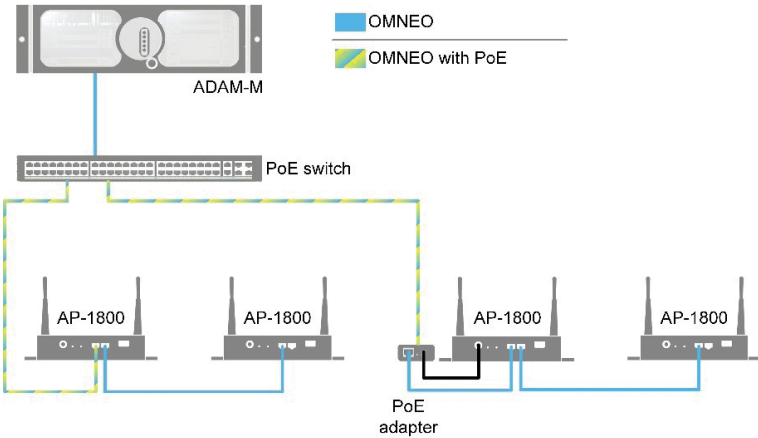


Figure 6. Connecting another AP to a PoE-enabled AP

CONCLUDING REMARKS

ROAMEO Access Points can work with PoE, but they require an adapter, which must be purchased separately (e.g., it is not included). Each AP that requires PoE must be connected directly to a PoE switch. The total power of all PoE enabled APs must not exceed the total power of the PoE switch.

WHITE PAPERS

DECT

ROAMEO uses DECT (Digital Enhanced Cordless Telecommunications) as its RF protocol. DECT is a mature radio technology that allows multiple users to occupy RF carriers by means of TDMA (Time Division Multiple Access), which means each carrier is divided into multiple timeslots. In DECT, there are 24 timeslots in a 10 mS window, which makes a frame. There is an downlink path timeslot(s) for communication from an Access Point (AP) to a Beltpack (BP) and a uplink path (BP to AP) timeslot(s). The Intercom use of the DECT Protocol only allows a total of 20 timeslots to be used while the other 4 timeslots are used internally by the DECT protocol. Both transmission paths use either one-timeslot or two timeslots per BP, depending on the amount of audio data that must be passed (CODEC selection). To simplify the discussion of timeslots and BPs, we refer to one timeslot as a system that uses a single timeslot for the forward path and a single timeslot for the reverse path. The same methodology applies to the use of two timeslots. The ROAMEO system is unique in the industry as it has two settings for the use of these timeslots. In narrowband mode, one timeslot is used per Beltpack to carry audio and control information. In wideband mode, two timeslots are used per Beltpack to carry audio and control information.

In Europe and most parts of the world, there are 10 RF carriers available. In the US, Canada and most of North America there are 5 RF carriers available. Other variations existing, using between 5 and 10 RF carriers, depending on the radio frequency rules and laws in those countries. Negotiation of the DECT frequency being used at any point in time is handled by the DECT protocol.

FREQUENCY MANAGEMENT

Reuse of frequencies (RF Carriers) can increase the number of BPs used over a wide area. Regulatory authorities for a country set the number and frequency of RF carriers that are available. Because of this, the total number of Beltpacks and Access Points in a single physical area of coverage is limited, regardless of the manufacturer of the DECT wireless intercom system. However, with the proper layout of a system, over a wide enough area, the reuse of RF carriers can occur. Thus, the number of Beltpacks over a wide physical area can exceed the number of BPs that a small single physical area can support. For example, consider Figure 1.

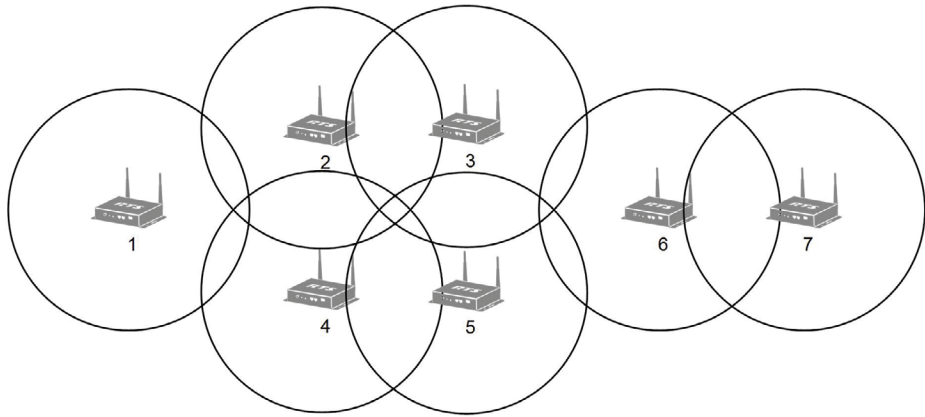


Figure 1. Frequency Reuse Situation Involving Seven US Frequency, Narrowband, Access Points and 56 BPs.

In this example in Figure 1, there are 7 US Frequency narrowband Access Points, each circle indicating the coverage area of each Access Point, respectively. In a given system, if each AP supports 8 BPs, then the whole system is supporting 56 BPs. This is above even the theoretical limit of what the 5 carriers in the US can support, but this is possible due to frequency reuse. No one area exceeds the 40 maximum BPs limit that RTS recommends (See Table 1) and it is below the 50 BP theoretical limit for a narrowband US Frequency system.

This is best shown by example. At AP 5, there is the potential that the adjacent BPs for AP's 2, 3, 4, and 6 could be detected by AP 5 if the BPs are all at or near the overlap coverage areas. AP 5 will then tag these BP timeslots as unavailable for use by BPs within AP's 5's coverage area. Therefore, the 32 timeslots taken by the

BPs of APs 2, 3, 4 and 6 plus AP 5's own 8 BPs timeslots is a total number of 40 timeslots. This number of timeslots is at the RTS recommend BP limit of 40 for a narrowband system, US Frequency system, so this area would be correctly balanced. However, if there were additional APs with a coverage area's intersecting AP 5 such that AP 5 could detect those BPs when they are in the overlapped coverage areas, then the capacity of this area would be exceeded and not enough BP timeslots would be available for all of the potential BPs in the area to allow intra-AP timeslot negotiation. This would then result in unstable operation of the system with Beltpacks dropping out or not being able to fully boot due to the lack of available timeslots in the area. All DECT based systems operate with these same limitations.

Looking at another area with the same operating conditions (Figure 1), AP 6 could detect the BPs of APs 3, 5 and 7, in the worst case scenario, when all BPs are near or at the overlap area of AP 6's coverage area. This would result in AP 6 detecting 24 BP timeslots as used. When added to AP 6's BP timeslots, that is a total of 32 timeslots being used, so this area is correctly balanced too.

Simplifying this concept, consider that each AP has, at any point in time, negotiated it's RF path frequencies and timeslots. The DECT Protocol, will avoid operating with frequencies and timeslots that are already used within the RF coverage area of an AP. The process of physically spreading out the APs and BPs to facilitate the reuse of frequencies can be used to make a RTS ROAMEO system of hundreds of BPs and many APs as shown in Figure 2. This can be done as long as the Access Points and Beltpacks are sufficiently spaced (RF spacing) from each other. The evaluation of an adjacent system's Access Points has to be done with respect to a worst case scenario which is at or below the recommended number of BPs that can be supported.

Figure 2 presents a more complex installation where there are 16 European frequency, narrowband APs covering studios in a media production center campus. Studios 1 – 7 are larger studios that have two APs each to support the number of users in each studio and the APs are in close proximity to each other, providing nearly the same coverage as shown. Studios 8 and 9 have fewer users and are smaller, so they only require one AP each. Table 1 indicates the recommended maximum number of Beltpacks for a European, narrowband system is 8 Beltpacks per Access Point and a total of 80 BPs in one area. This production center campus, with 16 APs, can use up to 128 Beltpacks without any issues due to frequency reuse. This particular installation may be using less BPs per AP than 8 due to timeslots being reserved for additional roaming BPs or they didn't fill out each AP fully in order to allow timeslots for the future BPs. However, the worst-case situation must be used to make sure the RTS recommend maximum capacity of Beltpacks in one area is not exceeded. For example, either AP in Studio 4 could potentially detect the BPs from Studios 1, 2, 3, and 5. With 8 BPs per AP, this equates to 64 BPs plus the 16 BPs from Studio 4 which would total 80 BPs. This system would be balanced, as the maximum number of BPs recommend per area is not exceeded. If you examine all the areas of Figure 2 you will find no one area exceeds the 80 BP limit. However, additional open timeslots may be necessary in a system where Beltpack roaming is required.

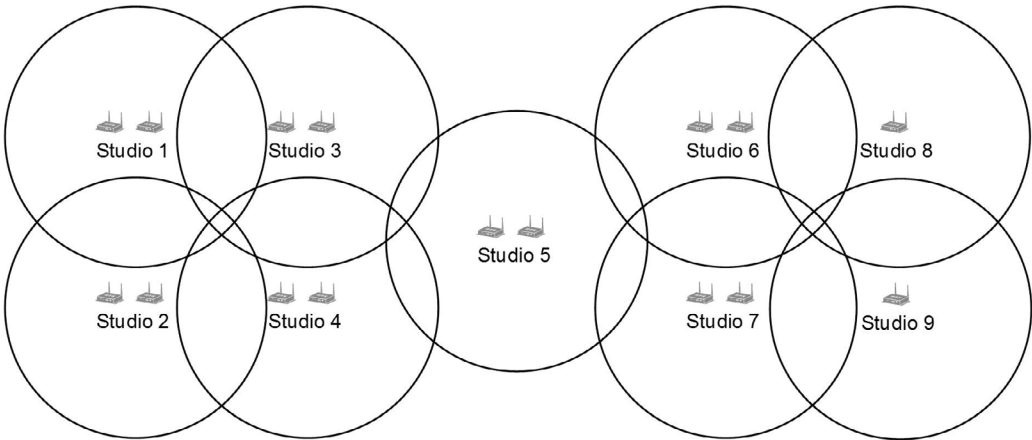


Figure 2. Frequency Reuse Situation Involving 16 European Frequency, Narrowband, Access Points and 128 BPs.

Table 1. RTS Maximum Limits Recommendations for a given System without Frequency reuse.

Area of World	System Mode	Maximum BPs per AP ¹ (AP Default Setting from Factory)	Number of RF Carriers (Regulatory Rules)	Max BPs an AP can Detect (Max Recommended)
Europe (Most of World)	Narrowband	8	10	80
Europe (Most of World)	Wideband	4	10	40
North America	Narrowband	8	5	40
North America	Wideband	4	5	20

¹See “ROAMEO Beltpack Capacity” White Paper to understand why a maximum number of Beltpacks per Access Point are recommended.

Reuse of frequencies (RF Carriers) can increase the number of BPs used over a wide area. Regulatory authorities for a country set the number and frequency of RF carriers that are available. Because of this, the total number of Beltpacks and Access Points in a single physical area of coverage is limited, regardless of the manufacturer of the DECT wireless intercom system. However, with the proper layout of a system, over a wide enough area, the reuse of RF carriers can occur. Thus, the number of Beltpacks over a wide physical area can exceed the number of BPs that a small single physical area can support. For example, consider Figure 1.

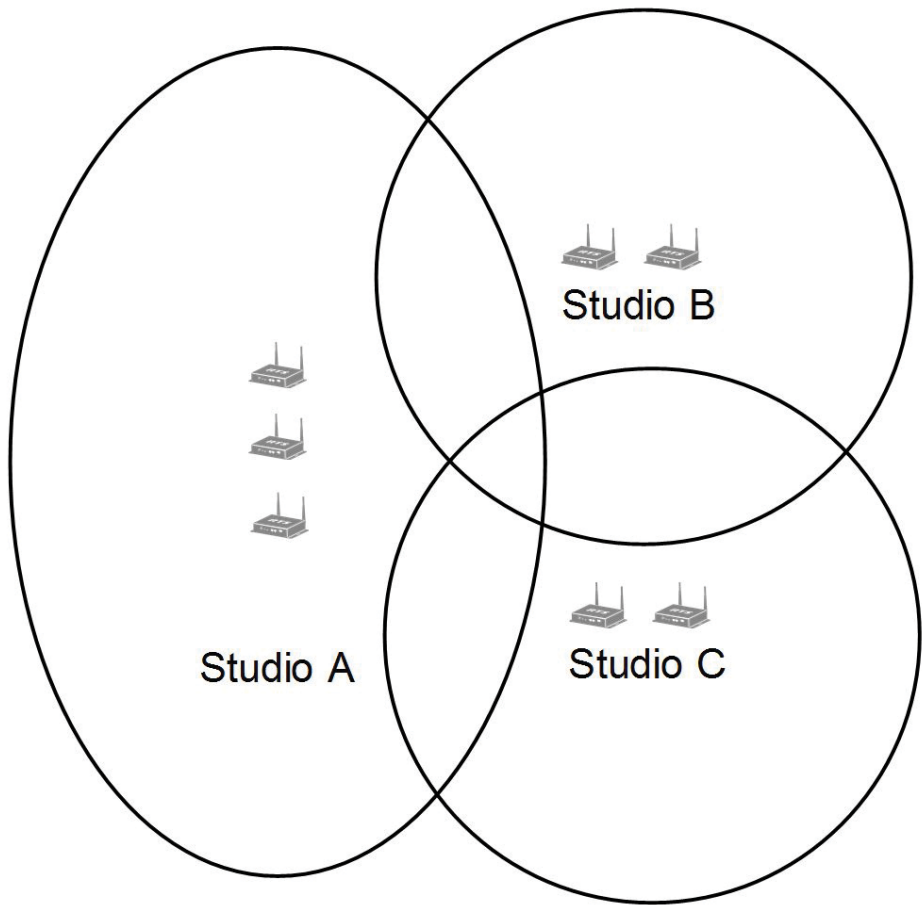


Figure 3. Frequency Reuse Failure Situation Involving 7 US Frequency, Wideband, Access Points and 28 BPs.

CONCLUSION

Through the proper layout of a system, taking advantage of the reuse of frequencies through physical spacing of APs and BPs, the number of Beltpacks in an installation can be increased well beyond the limit for a single physical area. Careful consideration should be taken in regard to the capacity of BPs that an area can support when designing systems that take advantage of frequency reuse. In Table 1, RTS recommends the BP limits that an AP can detect in any one physical area. The reuse considerations regarding RF spacing and DECT capacities apply to all DECT-based Systems.

Note that the examples herein are for systems that have the maximum number of Beltpacks per Access Point but still have ample timeslot availability for DECT intra-AP timeslot negotiation. This approach provides for the most stable systems. With respect to any DECT based Beltpack system, the maximum number of BPs that can be connected to an Access Point will vary greatly depending on the number of BPs that are desired to be roaming through a given area. Thoughtful planning should be performed to allow all Beltpacks to find timeslot availability, either while roaming or stationary, on one or more Access Points in a system.

Written by Jim Andersen and Kevin Moore, Bosch / RTS 2018



DECT

ROAMEO uses DECT (Digital Enhanced Cordless Telecommunications) as its RF protocol. DECT is a mature radio technology that allows multiple users to occupy RF carriers by means of TDMA (Time Division Multiple Access), which means each carrier frequency is divided into multiple timeslots. Each of these timeslots carries audio and control information for the system.

In Europe, and most parts of the world, there are 10 RF carriers available. In the United States, Canada and most of North America there are 5 RF carriers available. In South America, it is a mix between 5 and 10 RF carriers.

TIMESLOTS

Although there are 24 timeslots available (12 downlink and 12 uplink timeslots), the Intercom use of the DECT protocol only allows the use of a total of 20 timeslots. The other 4 timeslots are used internally by the DECT protocol. There are two types of wireless intercom DECT based products; products that use one timeslot and products that use two timeslots to convey audio and control information to a single Beltpack. Those companies that use one timeslot, typically, can support up to 10 Beltpacks per RF carrier but at the expense of some audio bandwidth or performance. The companies that use two timeslots can typically support up to 5 Beltpacks per RF carrier but with increased bandwidth for better audio performance. The ROAMEO system allows the user to decide. A user can choose more Beltpacks by selecting a specific CODEC in the configuration software, selecting narrowband operation or wideband operation. The narrowband CODEC uses one timeslot for each Beltpack and wideband CODEC uses two timeslots per Beltpack allowing for enhanced audio fidelity.

MAXIMUM CAPACITY

Once the CODEC configuration is known, the number of BPs supported per carrier frequency is also known; 5 BPs per carrier for systems that use two timeslots and 10 BPs per carrier for systems that use one timeslot. Given the number of RF carrier frequencies available and the CODEC selection, the maximum capacity of Beltpacks in an area is calculated by multiplying the two numbers as shown in Table 1 and 2:

Table 1. RTS Maximum Limits Recommendations for a given System without Frequency reuse.

Number of Beltpacks Per RF Carrier	Number of RF Carriers (Regulatory Rules)	Max. Beltpack Capacity of Area ¹
10 (Narrowband)	10	100
5 (Wideband)	10	50

Table 2. United States & Canada Maximum Capacity of a Physical Area

Number of Beltpacks Per RF Carrier	Number of RF Carriers (Regulatory Rules)	Max. Beltpack Capacity of Area ¹
10 (Narrowband)	5	50
5 (Wideband)	5	25

While some manufacturers may tell customers that the Tables are the Beltpack capacity that they can have in an area for their system, it is actually just the mathematical maximum. In real world situations, and as proven out by our testing as well as the DECT specification, the maximum capacity of devices is reduced by interference in the local environment and by adjacent Access Points (APs) and Beltpacks (BPs) use of the RF spectrum.

INTRA-ACCESS POINT (AP) FREQUENCY AND TIMESLOT NEGOTIATION

At maximum area capacity, as in Tables 1 & 2, there are no timeslots open on a carrier that allow a Beltpack to change frequencies or timeslots when interference is detected.

For robust systems, timeslots on each AP should be reserved for intra-AP frequency and timeslot negotiation. This provides for a better, interference free, optimized, connection on one of the RF carriers and timeslot(s) available in the area. DECT is a make-before-break audio protocol that requires that an open timeslot is confirmed before it will disconnect from the current timeslot. For effective intra-AP frequency and timeslot negotiation in your DECT system, please see Table 3 & 4 for the RTS Maximum Beltpack Capacity Recommendation.

Table 3. European (Most of World) RTS Maximum Beltpack Capacity Recommendations.

Max.. Beltpacks per Access Point ² (Access Point Default Setting from Factory)	Number of RF Carriers (Regulatory Rules)	Max. Beltpacks Over an Area ² (Max. Recommended for Effective Intra-AP Roaming)
8 (Narrowband)	10	80
4 (Wideband)	10	40

Table 4. U.S. & Canada (Most of North America & some of South America) RTS Maximum Beltpack Capacity Recommendations.

Max.. Beltpacks per Access Point ² (Access Point Default Setting from Factory)	Number of RF Carriers (Regulatory Rules)	Max. Beltpacks Over an Area ² (Max. Recommended for Effective Intra-AP Roaming)
8 (Narrowband)	5	40
4 (Wideband)	5	20

INTER-ACCESS POINT (AP TO AP) ROAMING

In order for Beltpacks to roam, DECT timeslots in a system (multiple Access Points) must be left open (unused by Beltpacks) to allow a BP to roam into an Access Point. How many DECT timeslots should remain open depends on the situation, but this should be considered as a necessary component of a successful system design.

SYSTEM SCALABILITY

The ROAMEO system is highly scalable. With the assertion that DECT RF frequency and timeslot management is being considered in the system design and, thus, effective RF spacing of reused DECT frequencies is being maintained for both Access Points and Beltpacks, then the effective ROAMEO system size (number of APs and number of BPs) can grow well beyond 80/40 Beltpacks using the narrowband or wideband CODEC. This will be covered in a separate RTS White Paper on DECT System Frequency Management. With careful system design, including radio frequency reuse management, the ROAMEO system is scalable beyond 100/50 Beltpacks in Europe and the United States, respectively.

CONCLUSION

When planning the number of Beltpacks that can be used in an area, do not just consider the maximum devices possible in a physical location. Consider that interference, such as DECT phone operation within the same area, which could occur. The system will need to have the ability to automatically adjust and optimize the RF paths. In order to optimize operation, the DECT protocol requires some empty timeslots to be reserved for these situations. In our recommendation, we account for these situations by reserving at least one open DECT timeslot on our Access Points for intra-AP frequency and timeslot negotiation for a system using the wideband CODEC and 2 timeslots for a system using the narrowband CODEC. In addition, at least one open DECT timeslot on an Access Point is needed for Access Point to Access Point (inter-AP) roaming. The ROAMEO system can support 100 Beltpacks in Europe and 50 in the United States. We recommend the maximum number of Beltpacks outlined above in Tables 3 and 4 to allow for effective DECT frequency and timeslot negotiation³. However, with proper frequency reuse management through effective Access Point and Beltpack spacing, the RTS ROAMEO system can easily scale to beyond 100/50 Beltpacks.

Written by Jim Andersen and Kevin Moore, Bosch / RTS 2018

¹ Maximum capacity of an area shouldn't be construed as recommended capacity. A system with a maximum capacity of Beltpacks (any manufacturer's DECT product) will not allow Beltpacks to roam freely between Access Points. Effective roaming of any manufacture's Beltpacks using DECT protocol requires both free timeslots (for both Beltpack Roaming and Intra-AP Access Point timeslot / frequency management) and non-overlapping frequencies. See Intra-AP timeslot section.

² The RTS AP-1800 Access Points can support up to 10 Beltpacks using one timeslot per Beltpack (narrowband mode) and 5 Beltpacks using two timeslots per Beltpack (wideband mode) but these numbers are reduced in the default setting from factory to make available frequencies and timeslots for intra-AP (within an Access Point) negotiation. The default number timeslots reserved for intra-AP frequency and timeslot negotiation per Access Points are narrowband = 2 and wideband = 1.

³ This DECT system limitation is the same for all manufactures of DECT-Based Beltpacks.

MATRIX & PARTYLINE

INTRODUCTION

Two-wire partyline is a simple and cost-effective way of providing communications to a number of users who can share one or two channels. As the name implies, this kind of system ties everyone together with a single cable. The cable goes into a beltpack, which also has a jack for a headset. Each beltpack has a cable to the next beltpack in the chain.

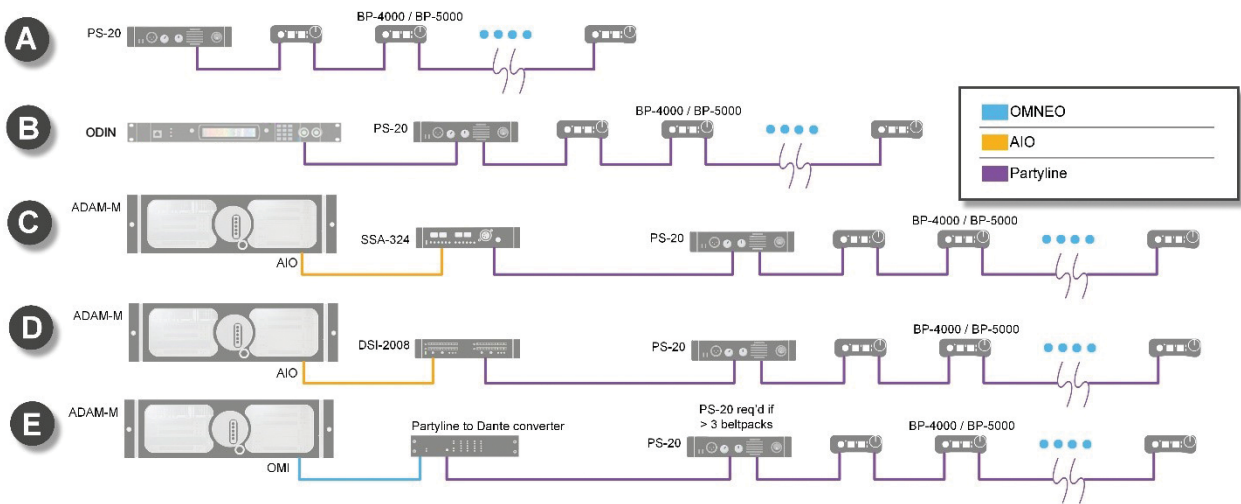
In a single-channel system, everyone speaks and listens to everyone. In a two-channel system, it is possible to create two separate groups. Two-wire partyline actually uses three wires. The wire carries audio, supply voltage, and ground. In a two-channel system, the wire carrying the supply voltage also has a signal superimposed. This signal scheme is usually referred to as ground-referenced. Systems with so-called differential signaling require additional wires but we will not get into that here.

STAND-ALONE VS MATRIXED OPERATION

A two-wire partyline system can be used with nothing except a power supply and multiple beltpacks. This is the stand-alone mode of operation. In many cases, it is convenient to connect a partyline to a matrix. That mode of operation provides additional possibilities but in most cases it requires an interface between the two-wire system and the matrix.

USE CASES

Figure 1. Five use cases for two-wire partyline



USE CASE A: STAND-ALONE

This is the simplest case, as discussed above. It requires nothing except beltpacks and a power supply. The example uses a PS-20 which is specific to the ground-referenced scheme from RTS.

USE CASE B: INTEGRATION WITH ODIN

A robust digital matrix in a compact 1RU space, built in OMNEO technology, eedundant power supplies, and energy-efficient design, uses less than 50W of power



Figure 2. Front view of ODIN

USE CASES C AND D: INTEGRATION WITH ADAM / ADAM-M

ADAM and ADAM-M are modular systems with cards that provide different capabilities. The AIO-16A card is used for connecting analog keypanels. Because the audio uses four electrical wires (plus two for keypanel data), this format is sometimes called four-wire. Two-wire and four-wire cannot be inter-connected without an adapter. RTS has two, the SSA-324 (an analog device) and the DSI-2008 (which has digital signal processing). Figure 3 and Figure 4 show the back of these two converters. The user manuals for these two products explain how the electrical connections must be made.



Figure 3. Back of the SSA-324



Figure 4. Back of DSI-2008

USE CASE E: INTEGRATION VIA OMNEO

It is possible to go directly from a two-wire partyline to high-quality audio over IP. Dante is supported by the OMNEO Matrix Interface (OMI), a card that goes into both ADAM and ADAM-M. Converters are available from several third-party suppliers. This example uses the M45DR from Studio Technologies. Figure 5 shows the back of that device.



Figure 5. Back of the StudioTech M45DR

The M45DR communicates via the RJ45 connector on the rear. In most cases, it would be connected to an Ethernet switch, which in turn is connected to the OMI-card. Dante eliminates the problem of creating unique analog cables, to connect from one device to another, as would be required in use cases C and D.

CONCLUSION

Two-wire partyline systems can be used in a stand-alone or matrixed configuration. The use cases are summarized in the table below.

Configuration	Matrix	Matrix Card	Converter
Stand-alone	N/A	N/A	N/A
Matrixed	ODIN	N/A	N/A
Matrixed	ADAM / ADAM-M	AIO-16A	SSA-324 or DSI-2008
Matrixed	ADAM / ADAM-M	OMI	M45DR

BASIC USE CASE

RVON is a way of getting audio across long distance, without the need for dedicated infrastructure. RVON encodes the audio to allow it to be sent over the Internet. To use RVON, a matrix card is required, as well as a device on the other end, capable of sending and receiving RVON audio. This is shown in Figure 1.

With the exception of one single keypanel, the KP-3016A, **all** keypanels in the KP-Series family are capable of running RVON. The firmware for RVON may be downloaded for free from the RTS website.

USING THE RVON-I/O

For analog products that need to be connected remotely, there is a converter called RVON-I/O. Using the RVON-I/O, it is possible to connect analog products as well as legacy products, as shown in Figure 2.

Figure 2 shows RVON being used to connect an IFB, a KP-Series keypanel connected via AIO, a legacy keypanel (KP-96/7) and a Zeus III with an analog keypanel. There is also a software product, the Virtual Keypanel or VKP, running on a PC, which gives it the ability to transmit and receive RVON audio. Note the Cronus, ADAM, and ADAM-M all have the ability to use RVON, whereas Zeus III does not.

RVON IN TRUNKING APPLICATION

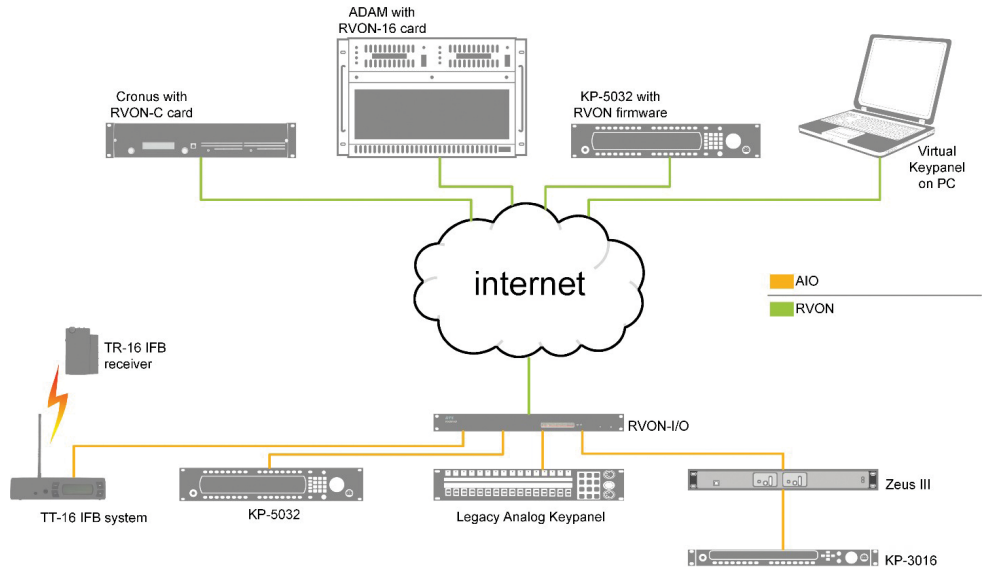
One of the most commonly used applications for RVON is Trunking. An example is shown in Figure 3. Three intercoms are interconnected over RVON. The Trunkmaster issues commands to the controllers in each of the matrices to close crosspoints, meaning set up paths for audio. It is very important to understand the Trunkmaster does not actually switch any audio.

Figure 3. RVON used in Trunking

Figure 1. RVON in its simplest form



Figure 2. Using the RVON-I/O box for remoting analog devices

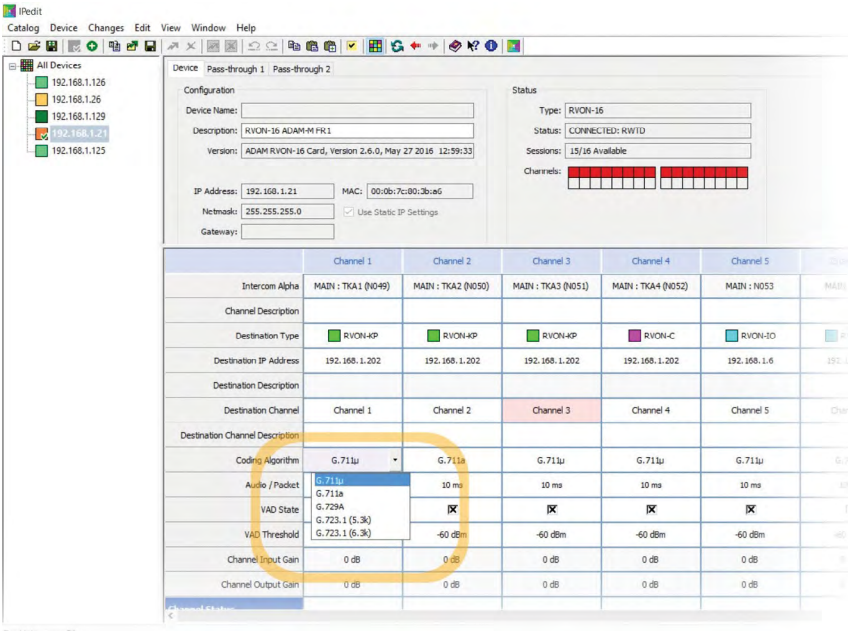


RVON FOR REDUCED BITRATE

RVON is not one single codec, it is three. The RVON hardware from RTS supports three codecs, G.711, G.729, and G.723 – all with different technical characteristics and levels of compression, as summarized in Table 1.

Codec	Bit Rate	Coding Delay	Playout Delay	Bandwidth	Sample Rate
G.711	64 kbit/s	0.125 ms	20-60 ms	160-224 kbit/s	8k
G.729	8 kbit/s	10 ms	20-120 ms	32-112 kbit/s	8k
G.723	5.3/6.3 kbit/s	30 ms	60-120 ms	29-45 kbit/s	8k

Table 1. RVON has three user-selectable codecs



The codec can be configured in IP-Edit. An example is shown in Figure 4.

Figure 4. Codec can be selected in IP-Edit

RVON PRODUCTS

RTS has multiple RVON products to support the use of RVON in virtually any application:

- Matrix cards for Cronus, ADAM, and ADAM-M
- Firmware for KP-Series of keypanels (firmware may be downloaded for free)
- Software for ODIN
- Option cards for KP-32, RP-1000, KP 12 CLD, and DKP 16 CLD
- Software for PC (Virtual Keypanel)
- RVON-I/O converter for analog

Finally, a couple of technical points:

- RVON requires fixed IP-addresses to work. The network cannot support Network Address Translation (NAT). In networks that do support NAT, there are still solutions for making RVON work, but they may require additional hardware devices (to create VPN tunnels).
- The RVON firmware for KP-Series keypanels may be downloaded for free. To install it, an RTS supplied software program called Firmware Upload Tool (FWUT) must be used. FWUT is included with the OMNEO Suite, which may also be downloaded from the RTS website.

PROBLEM/SOLUTION

AES67 is an open public standard that defines the use of technologies to allow the exchange of audio and clocking information between devices that implement a variety of different IP media networking solutions. Intended to increase interoperability between different media networking technologies currently on the market today, AES67 allows the exchange of audio among different technologies and vendors, an application that previously required expensive and complicated protocol conversions and configuration. However, as AES67 defines a limited subset of media networking features, it is best implemented inside of a more complete media networking architecture such as Dante, Ravenna, LiveWire or other technologies that support AES67 interoperability. Due to the different implementations of AES67, and media networking elements that are not included in the AES67 specification (such as discovery methods or connection management), some additional details are needed to properly configure the exchange of audio between these different systems and technologies. Though there are many technologies and implementations of AES67 on the market today, this application note addresses the details of connecting two specific implementation - OMNEO, which uses Audinate’s Dante, and Ravenna, which is used by some manufacturers, such as Lawo.

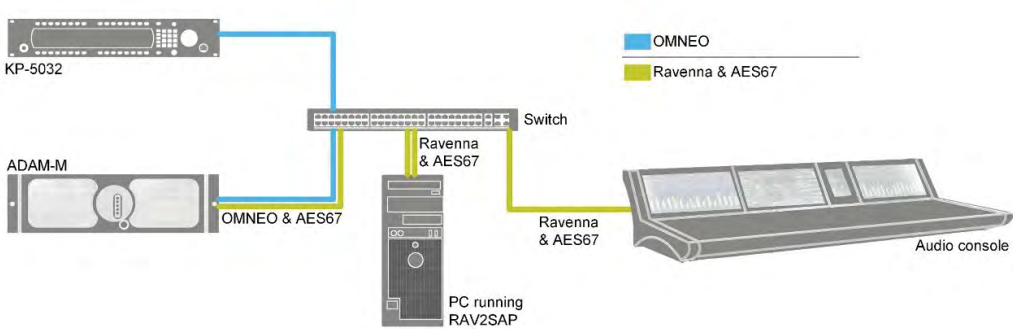


Figure 1. Connecting OMNEO devices to Lawo

TECHNICAL CONSIDERATIONS

When preparing to interface a RTS OMNEO product to a Ravenna product, it is important to consider the following details about how AES67 will work within the installed system.

Discovery - AES67 defines multiple discovery mechanisms that may be implemented. In order to work with OMNEO or other Dante devices, Session Announcement Protocol (SAP) must be used. This enables Dante Controller to see the other AES67 devices natively so the user can configure connections. Though Ravenna implements a unique discovery mechanism, ALC NetworX provides a free utility called RAV2SAP that is able to convert these native advertisements to SAP so that Dante Controller may see the advertisements to establish connections. This is shown in Figure 1. The PC shown in the figure must always be connected to the system.

Multicast audio only - Currently OMNEO and Dante devices that support AES67 only support the use of multicast AES67 audio streams. The multicast addresses that are used must be in the range of 239.69.0.0-239.69.255.255/16

Connections are not persistent after a system reboot - Because of the mechanisms that are used for connection management, established connections are not retained between system power cycles. Additionally, a computer running the RAV2SAP software must be left online to maintain the discovery advertisements. If a system is power cycled, these connections will need to be manually reestablished.

Fixed 2 ms latency - Although technologies such as OMNEO and Dante support much higher latencies to allow greater distances, AES67 only allows 2 ms latencies. A manufacturer may implement additional options.

Clocking - AES67 requires the use of IEEE 1588 v2 clocking. This poses something of a challenge for the network as OMNEO and Dante also need to support the use of IEEE 1588 v1 to maintain compatibility with non-AES67 Dante equipment. Fortunately, AES67-enabled Dante and OMNEO devices support both IEEE 1588 v1 and v2, although care needs to be taken in the system configuration to ensure proper operation. The most straight-forward approach is outlined below, that of using an AES67-enabled Dante/OMNEO device as master for both PTP domains.

1. Enable the device’s “Preferred Master” status. In our example, that is the OMI-card of the ADAM-M matrix.
2. Disable “Preferred Master” status for all Dante/OMNEO devices that have AES67 disabled. In our example, that would be the KP-5032 keypanel.
3. Disable “Sync to External” for all devices.
4. Assign a PTP version 2 priority level between 128 and 255 for all non-Dante/non-OMNEO devices.

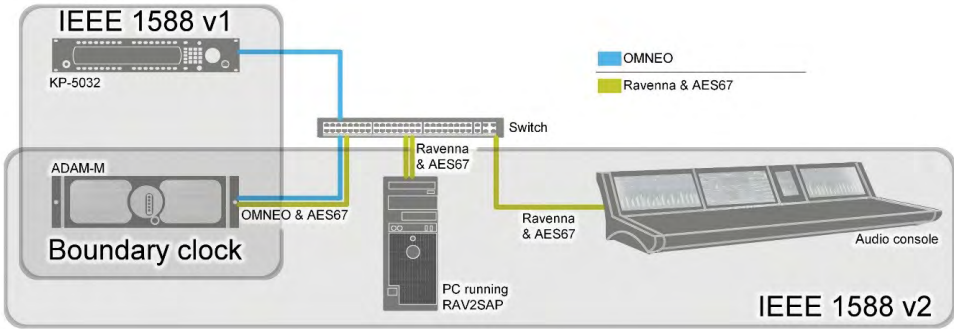


Figure 2. The OMI-card in the ADAM-M bridges the two PTP domains

QoS Settings - Dante and AES67 use different DSCP values for their audio and clocking packets. This can create a problem as priority conflicts may arise when QoS is configured on a managed network. However, these values can be remapped in a couple of different ways. Ravenna supports the remapping of DSCP values so these can be changed to match those needed by Dante. Alternately, these values can be remapped in a managed network switch that supports this feature.

PRACTICAL CONSIDERATIONS

AES67 provides a level of interoperability between IP media networking systems that has never been achieved before without extensive complexity and use of converters, although it does require additional configurations compared to a standard OMNEO system.

Use AES67 between systems - The primary benefit of AES67 is provided when it is used to connect different systems. As architected media networking solutions such as OMNEO offer additional functionality and features, AES67 is generally used to connect different systems as opposed to being the primary transport for all devices.

Consider network design carefully - Planning for the deployment of systems requires careful thought as to how to connect devices and where to use AES67 links. For example, OMNEO supports latencies up to 20 ms and these allow very long distances between devices. In a situation such as this, AES67 should be used on one side of the link or the other and OMNEO should be used for the long-haul connection.

Consult the manufacturer - AES67 can be implemented in different ways by different manufacturers. Be sure to consult the manufacturer for specific details on their implementation to ensure that elements related to discovery, clocking, QoS, audio formats, and other details will meet your needs.

PROBLEM/SOLUTION

Intercom systems often have a need to connect to phone systems. In the past, Plain Old Telephone Service, or POTS, was readily available from the local phone company, but with the migration to IP, old-style POTS lines are hard to get. For a while, ISDN lines seemed to be the future, but in most locations, even ISDN is disappearing. Session Initiation Protocol, or SIP, is the dominant IP-based protocol for telephony today. SIP is the protocol spoken by most private branch exchanges, or PBXs. The topic of this Application Note is connecting SIP phones to an intercom.

SIP-SOLUTIONS FROM RTS

RTS has hardware solutions that provide SIP-connectivity. However, we will look at the possibility of providing SIP through a software solution that requires very little additional hardware, and is highly scalable. Figure 1 shows the basic system diagram.

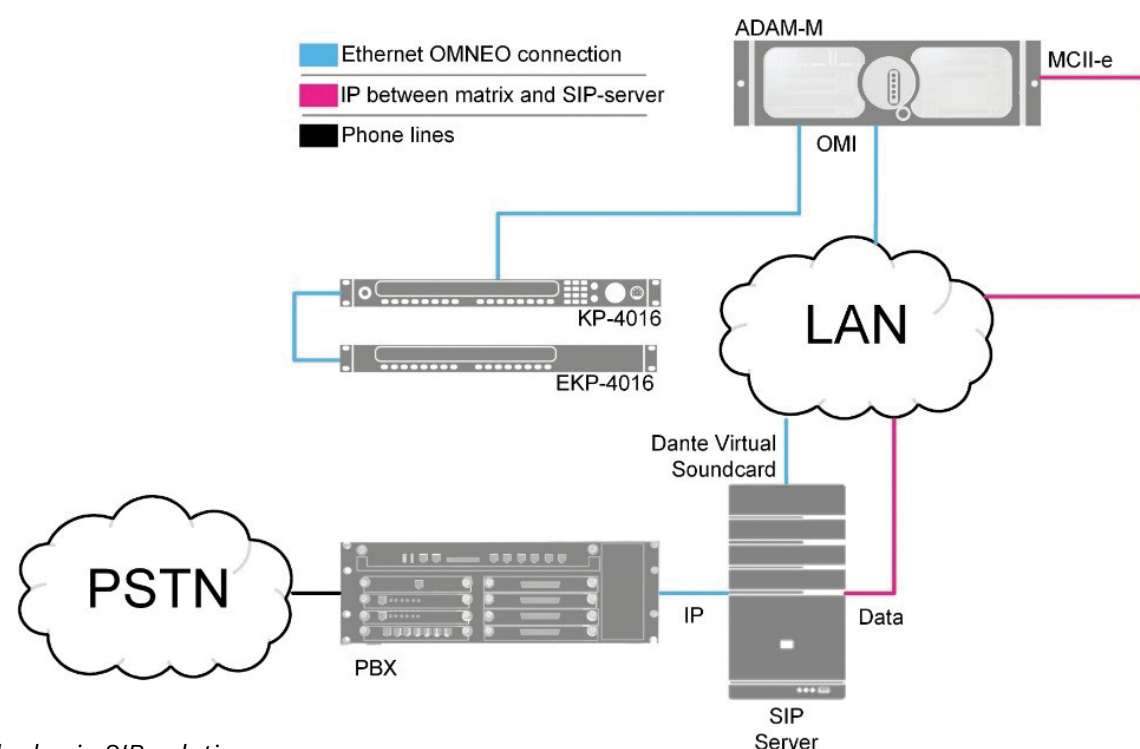


Figure 1. The basic SLP-solution

The matrix, an ADAM-M in this example, is connected to a dedicated PC running special software. The PC acts as a SIP Server. It has two pieces of software installed:

- Dante Virtual Soundcard from Audinate – this is the software device that sends audio to and receives audio from the matrix in the form of high-quality audio over IP. The ADAM-M has an OMI-card, which sends and receives OMNEO, the Bosch-branded version of Dante.
- VLINK software – this RTS software is responsible for handling the SIP communication itself. SIP is simply a software feature within the VLINK software. However, this is done in the licensing process, not customer setup.

Note that no other hardware devices are required. The master controller of the ADAM or ADAM-M matrix is able to speak directly to the VLINK software, over IP. The SIP-functionality is supported in the current master controller software (version 3.3.0 and later).

If the matrix does not have an OMI-card, it is possible to use an analog connection. However, since the SIP-Server is expecting digital audio over IP, it must be converted to Dante with an external conversion box as shown in Figure 2. The example uses a Focusrite RedNet 1 box.

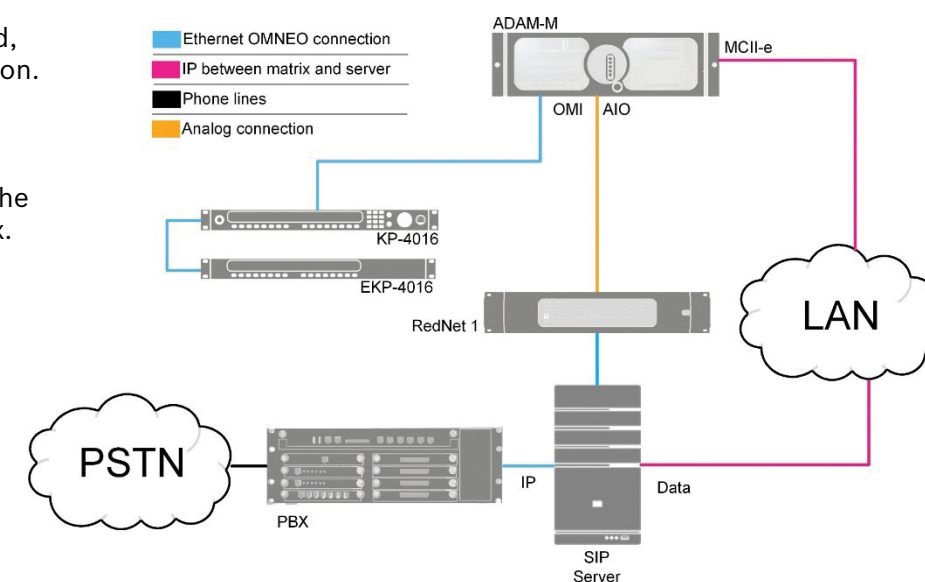


Figure 2. SLP-example using an analog output from the matrix

MIXED USE OF VLINK

As we have seen, VLINK is capable of being an interface to SIP. The traditional use of VLINK for intercom applications, which is still supported, is to run the software on a smartphone and have it work as a keypanel. This requires a separate server, as shown in Figure 3.

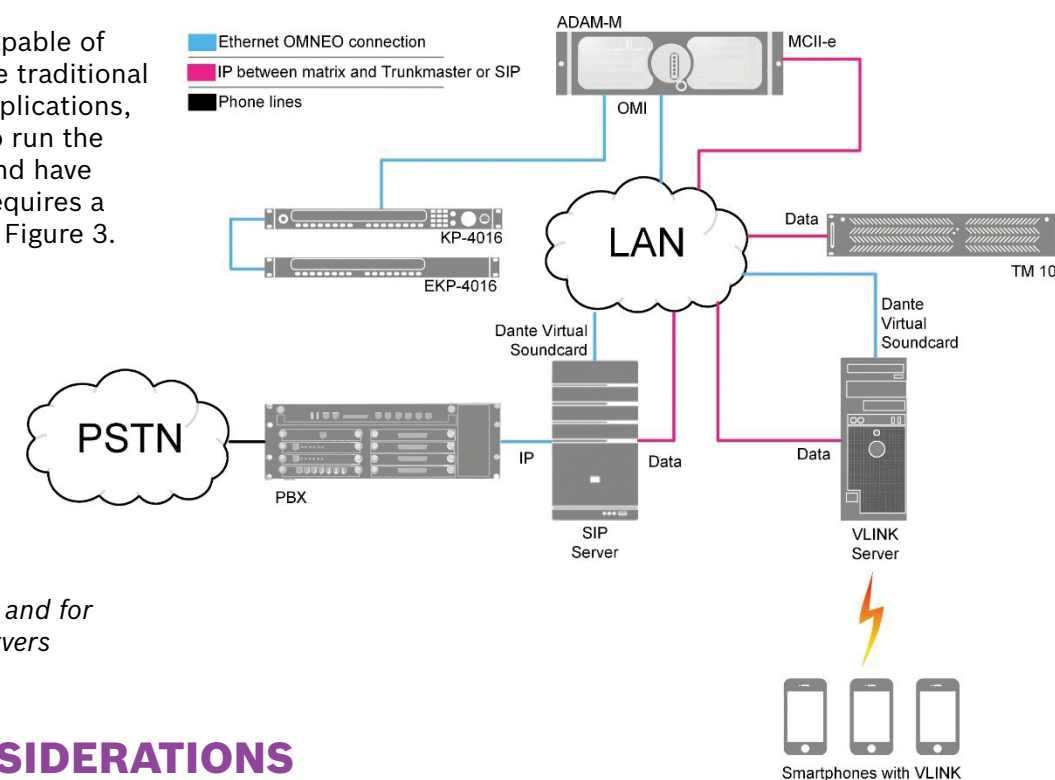


Figure 3. Use of VLINK for SIP and for Smartphones requires two servers

TECHNICAL CONSIDERATIONS

- SIP is available for the ADAM and ADAM-M matrices.
- The SIP-Server requires VLINK licenses, plus a Dante Virtual Soundcard.
- Each SIP line requires two VLINK ports, which must be licensed. Customers pay for one port only.
- SIP-Server does not require a Trunkmaster.
- When VLINK is used for SIP and for Smartphones, two separate servers are required.
- An OMI-card is recommended in the matrix. If analog audio is used, an external converter must be used.
- A software-based Tally-screen is available separately. It shows status of all SIP-lines.

INTRODUCTION

VLINK is the name of a product from RTS that provides matrix access from smartphones or connected laptops. With the VLINK app installed on a smartphone, it is possible to communicate with other users on the matrix system. The smartphone behaves like a portable keypanel. This is illustrated in Figure 1.



Figure 1. Matrix access on your smartphone

SOLUTION

A basic VLINK system is shown in Figure 2.

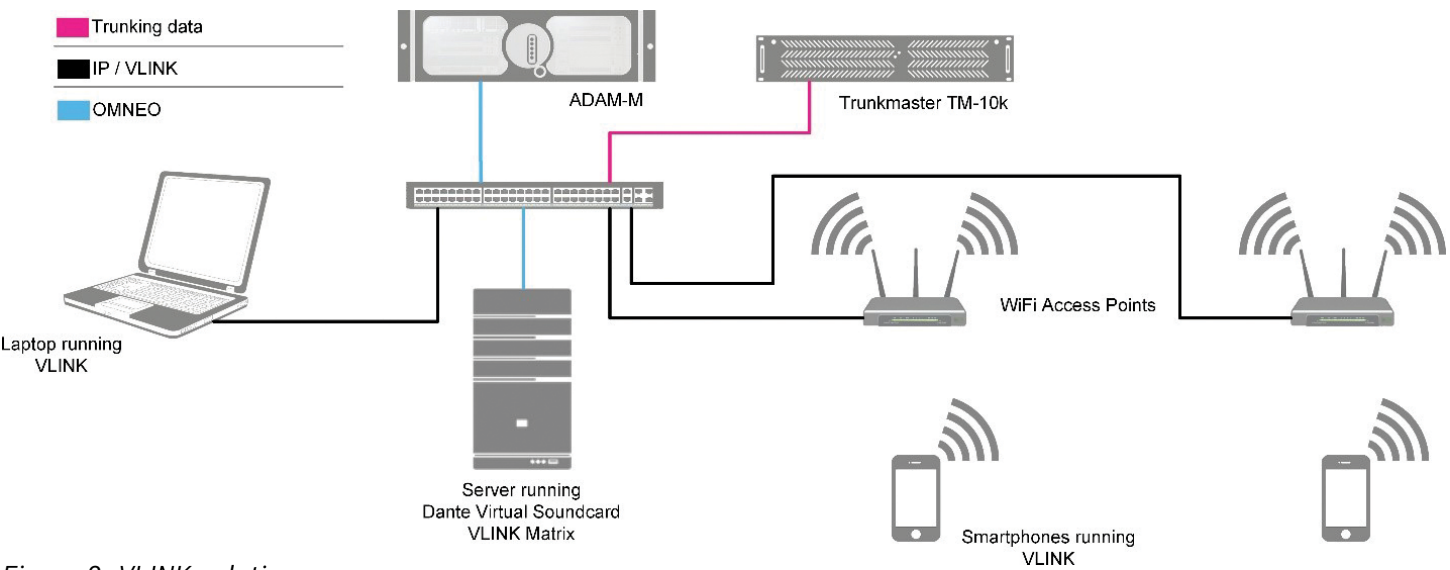


Figure 2. VLINK solution

The VLINK matrix software runs on a stand-alone computer. OMNEO is used to transmit audio between the matrix (an ADAM or ADAM-M) and the VLINK system. An OMNEO Matrix Interface (OMI) card gives ADAM-M the ability to use OMNEO. In the example, VLINK is also being used on a laptop. The computer running VLINK has Dante Virtual Soundcard installed, which is what allows the hardware to receive audio from any Dante-compatible device, in this case the OMI-card. Each connected device (two smartphones and a laptop, in this example) requires a valid license on the computer. The computer with the VLINK software behaves as a matrix, which is why the Trunkmaster TM-10k is required.



Figure 3. Screenshots of VLINK of various device types

VLINK SYSTEM ADMINISTRATION

VLINK is dynamically configurable from any workstation on the network using the intuitive web based VLINK System Administration application.

Administrators can:

- View system runtime, number of clients connected, active audio inputs and outputs, and audio statistics from all clients, and system-activity log
- View and change network settings, system sampling rate, and selector color schemes that indicate audio
- View, add, edit, and delete configured users and devices and log in names, passwords, selector labels and client type.
- Set system audio parameters and enable automatic gain control and echo cancellation per user or globally.

SIP INTEGRATION

VLINK also enables integration with SIP-based telephone systems. You can read more about that in the Application Note titled “SIP Server”. Integration with SIP-based telephone systems does not require a TrunkMaster TM-10k.

INTRODUCTION

Trunking is the use of interconnections between two or more intercom matrices, to enable a user on one intercom to talk to a user on another, as if they were on the same system. Audio signals can be transmitted from one matrix to another using a variety of technologies such as AIO, OMNEO, MADi, or RVON (RTS Voice Over Network, a VoIP technology). Figure 1 shows an example that uses AIO audio only to interconnect three ADAM-M matrices.

AIO, OMNEO, and MADi all require physical connections, which of course costs money. If a user on the upper matrix is trying to talk to a user on the lower left matrix, that works provided at least one connection is available. If both are busy, there is no way of getting through. It's potentially poor use of resources, too. The system has six trunk lines, four of which could have provided an alternate path. The type of static trunking shown in Figure 1 is sometimes called “dumb four-wire” because it transmits audio only, using one pair of cables for audio in one direction, and one pair for the other.

If the system were smart enough, it could have routed the call via the matrix on the lower right. Trunkmaster is a device that will achieve exactly this. Figure 2 shows how it is connected in the system.

In this scenario, the call from our example will go through, provided there's at least one available connection from the top matrix to the lower right, and at least one from the bottom right to the left. Note the lines going to and from the Trunkmaster are data lines. The Trunkmaster does not transmit audio. It only instructs the matrices to close crosspoints, to enable a call to go through.

Although a more efficient use of transmission capacity is an advantage, the main advantage of trunking is the ability of the system to use the trunks for any system function. The Trunkmaster dynamically decides for which intercom function it wants to use the line. That is not possible in the scenario in Figure 1.

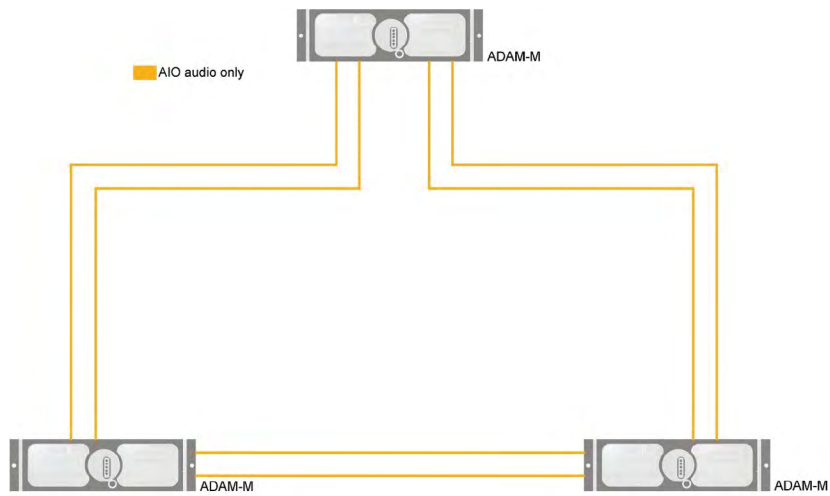


Figure 1. Trunk lines interconnecting three matrices

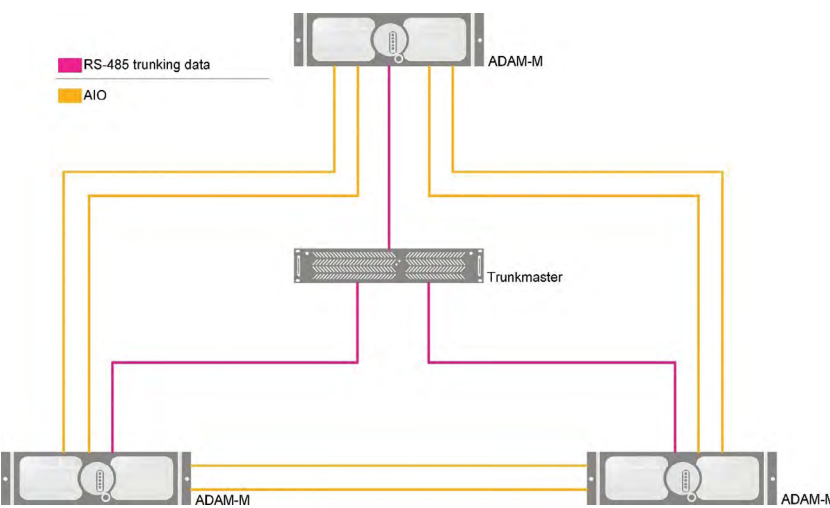


Figure 2. A three-matrix system with a Trunkmaster

The example above uses RS-485 serial data connections. Trunkmaster has a total of 32 ports for RS-485 data. If the matrix and Trunkmaster are co-located, an RS-485 connection may be the simplest way to create a data connection between the devices. RS-485 is independent of any IP network settings because it is a serial protocol, not based on IP. If a serial data connection is not practical, it is also possible to use an IP-based protocol, as shown in Figure 3.

The instructions from the Trunkmaster to the master controllers in each matrix are transmitted as IP-packages, so this architecture goes beyond 32 matrices. In fact, a single Trunkmaster can manage up to 255 matrices, each with 880 ports, for a total of 225,000 ports. By all accounts, a 32 matrix system is a very large system (up to 28,160 ports). Even in the largest trunking system in the world, more than 32 matrices are very rarely active at the same time.

The architecture shown above can be improved in a number of ways, and we'll look at three ways of making it more resilient.

USING A REDUNDANT OPTICAL FIBER RING

The architecture shown in Figure 3 is sensitive to individual physical connections going down or getting accidentally disconnected. Physical diversity, e.g., pulling the cables through different ducts, decreases the risk of simultaneous failure of two connections, but is not always easy to achieve. Figure 4 shows an architecture that uses the FMI-4 to create redundancy for each connection. Technically, the FMI is a multiplexer, but they are usually referred to as fiber conversion interfaces or fiber conversion boxes.

The optical fiber ring is highly resilient to faults. First, each green line is actually two fibers. Second, even if both fibers between any two multiplexors are severed or accidentally disconnected, the audio is automatically recovered within microseconds, because there is an alternate path. The optical fiber protocol is capable of carrying analog audio and digital data, in this case the trunking data. An optical fiber ring may only make sense if the matrices are relatively far apart.

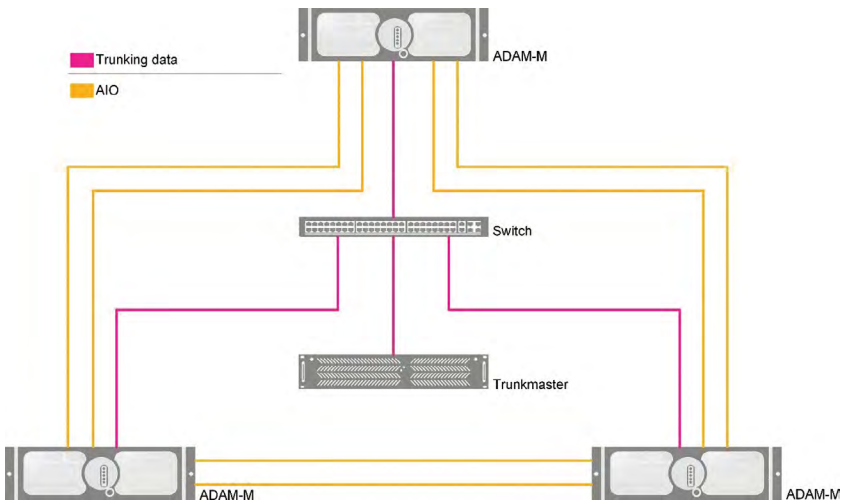


Figure 3. Trunking using IP for the data connections

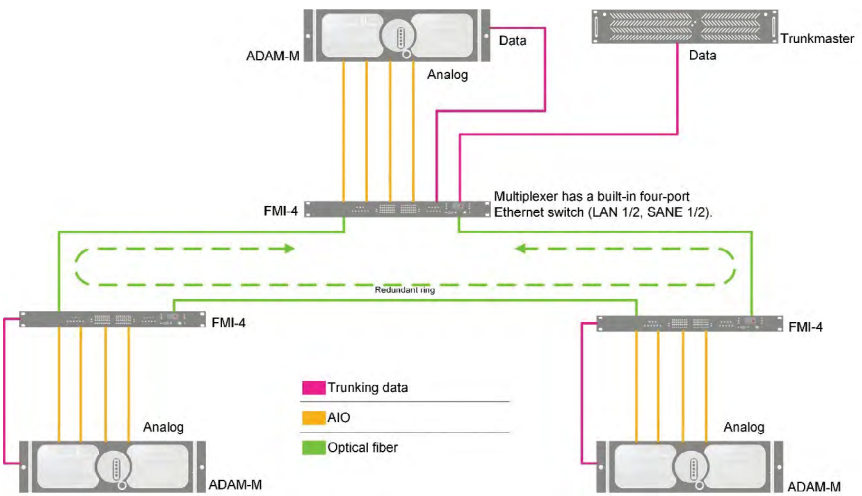


Figure 4. A redundant optical fiber ring creates a backup for each connection

USING RVON PROTOCOL OVER INTERNET

The architecture in Figure 4 requires the use of specialized multiplexors to create a very high level of resilience and a fully predictable and fixed latency. A slightly variable network latency (e.g., the delay of individual packets varies) can be accommodated by the RVON protocol, a VoIP protocol developed by RTS. The architecture in Figure 5 shows a solution that uses internet to interconnect the matrices.

The green lines are audio lines that use RVON. This requires the RVON-16 card in each of the matrices. The Trunkmaster data is also transmitted over internet. RVON technology applied to Trunking represents a very realistic use case. In fact, it is the most commonly used solution. Compared to an uncompressed analog signal, some audio quality is sacrificed, but as long as an internet connection is available, an audio path can always be established between the matrices.

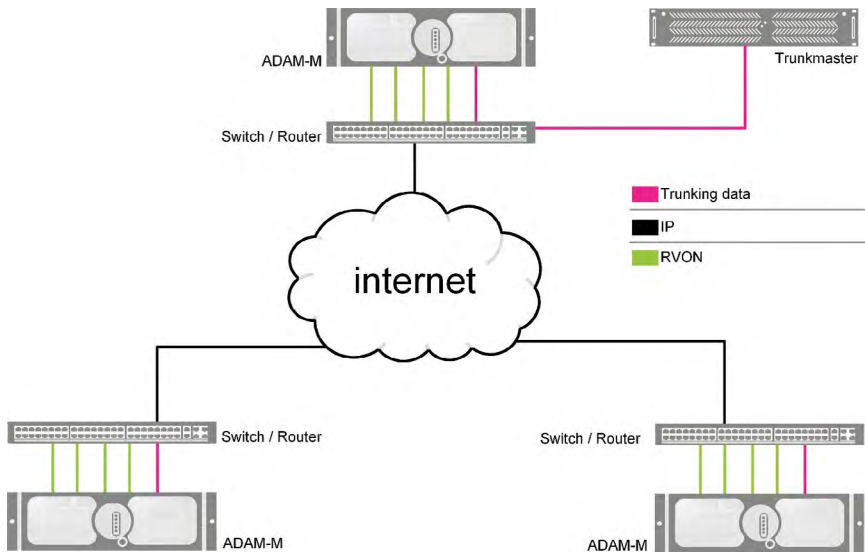


Figure 5. Trunking using RVON

REDUNDANT TRUNKMASTERS

In Figure 5, the Trunkmaster itself is still a so-called single source of failure. If it dies, the trunking network will stop working. Figure 6 shows a solution where two Trunkmasters are working in tandem. If one fails, control is shifted to the other. The device that makes this possible is the SWP-2000. The connection between the SWP-2000 and the two Trunkmasters uses a proprietary 25-pin cable. The graphic also shows two laptops connected to the SWP-2000, running two software packages for configuration and overview of the trunking network.

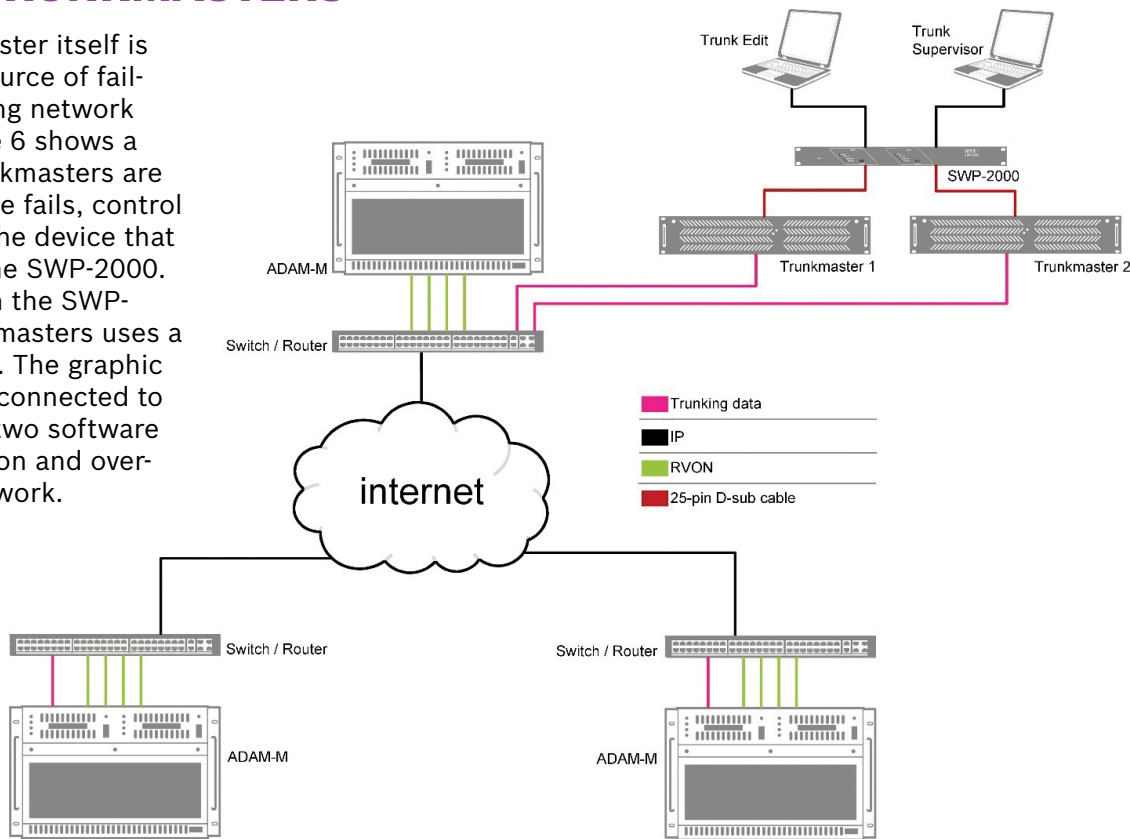


Figure 6. A solution that uses a redundant Trunkmaster

A REAL TRUNKED NETWORK

Figure 7 shows a real trunked network from a large broadcaster.

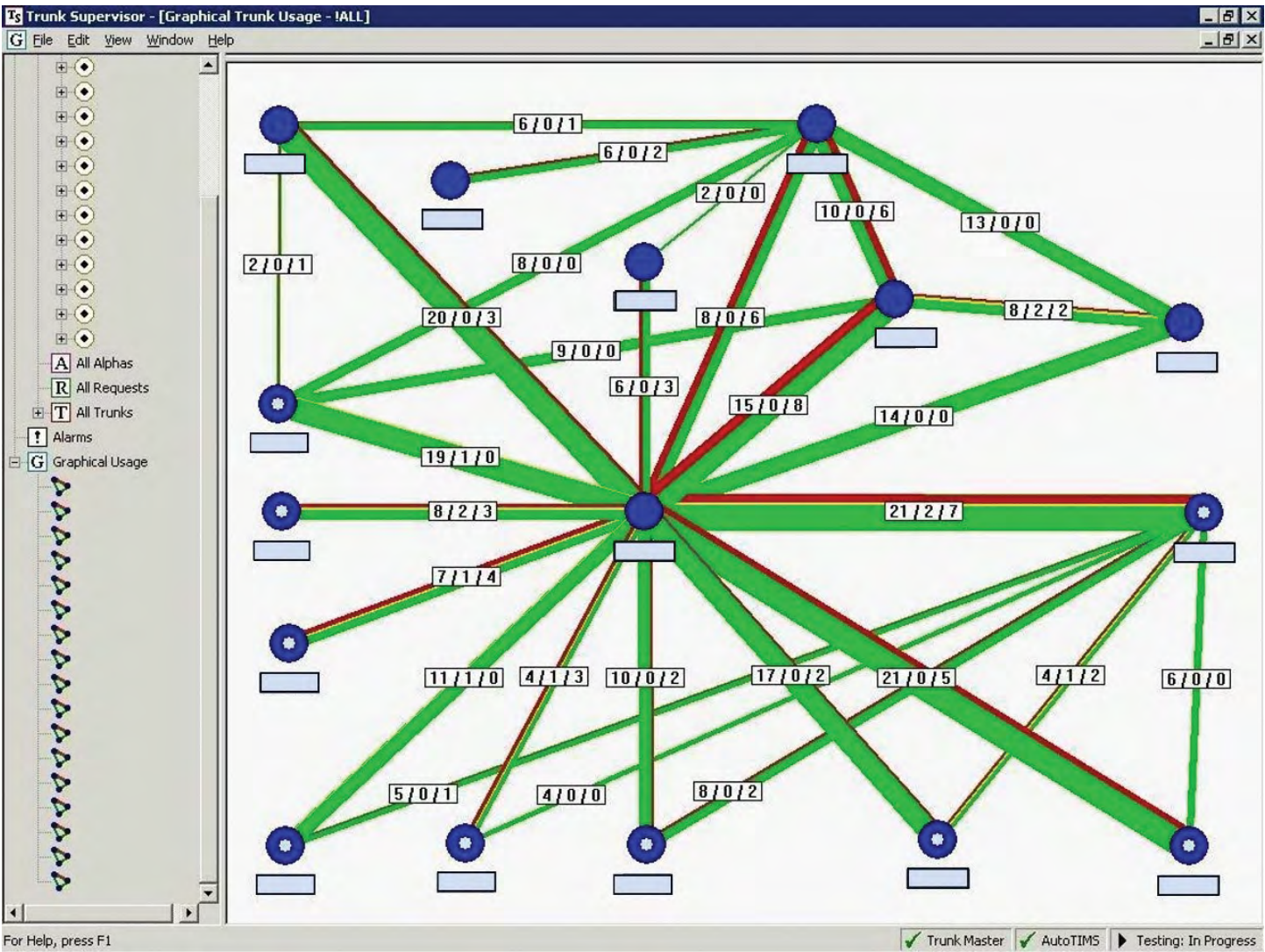


Figure 7. A large trunked intercom system

The thickness of the green lines indicate how many trunk lines are available, and the red is what's actually in use at this very instant. The names of the locations have been removed entirely from the overview, but they would appear in the blue boxes to identify the locations, as well as on the overview pane on the left hand side.

PROBLEM/SOLUTION

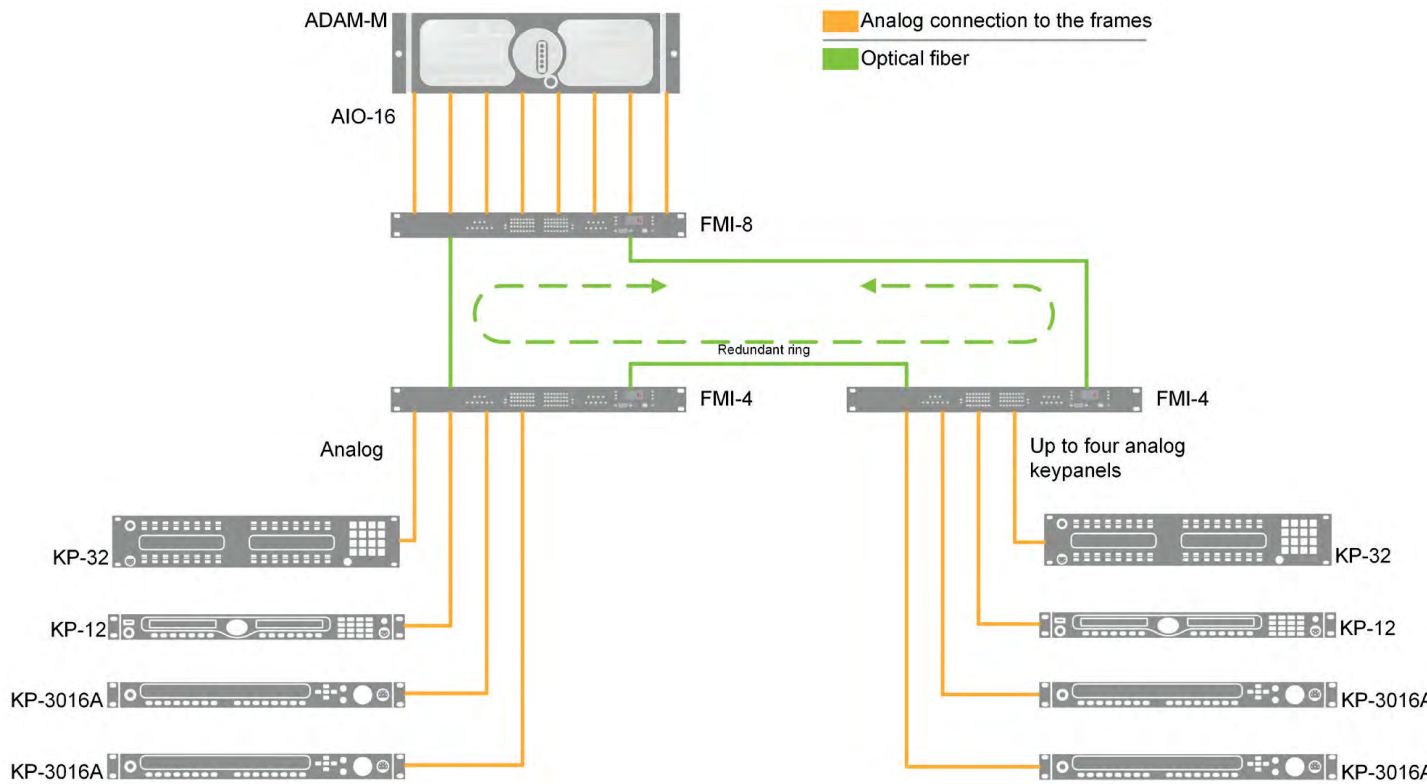
Keypanels are critical to live production. Backup connections for keypanels are essential in case a cable breaks. FMI multiplexers allow keypanels to communicate, even if a cable breaks or is accidentally disconnected. The multiplexers are interconnected with redundant optical fibers; if one fiber fails, the other takes over without interrupting the signal. Multiple multiplexers can be configured in a ring, which provides redundancy for keypanels in multiple locations.

The most common redundancy configuration is to have two multiplexers, connected to one another with two fibers. More complex configurations are possible. Multiple multiplexers can be chained together in a ring topology. A ring topology is resilient to a single fault anywhere on the ring.

PRACTICAL EXAMPLE

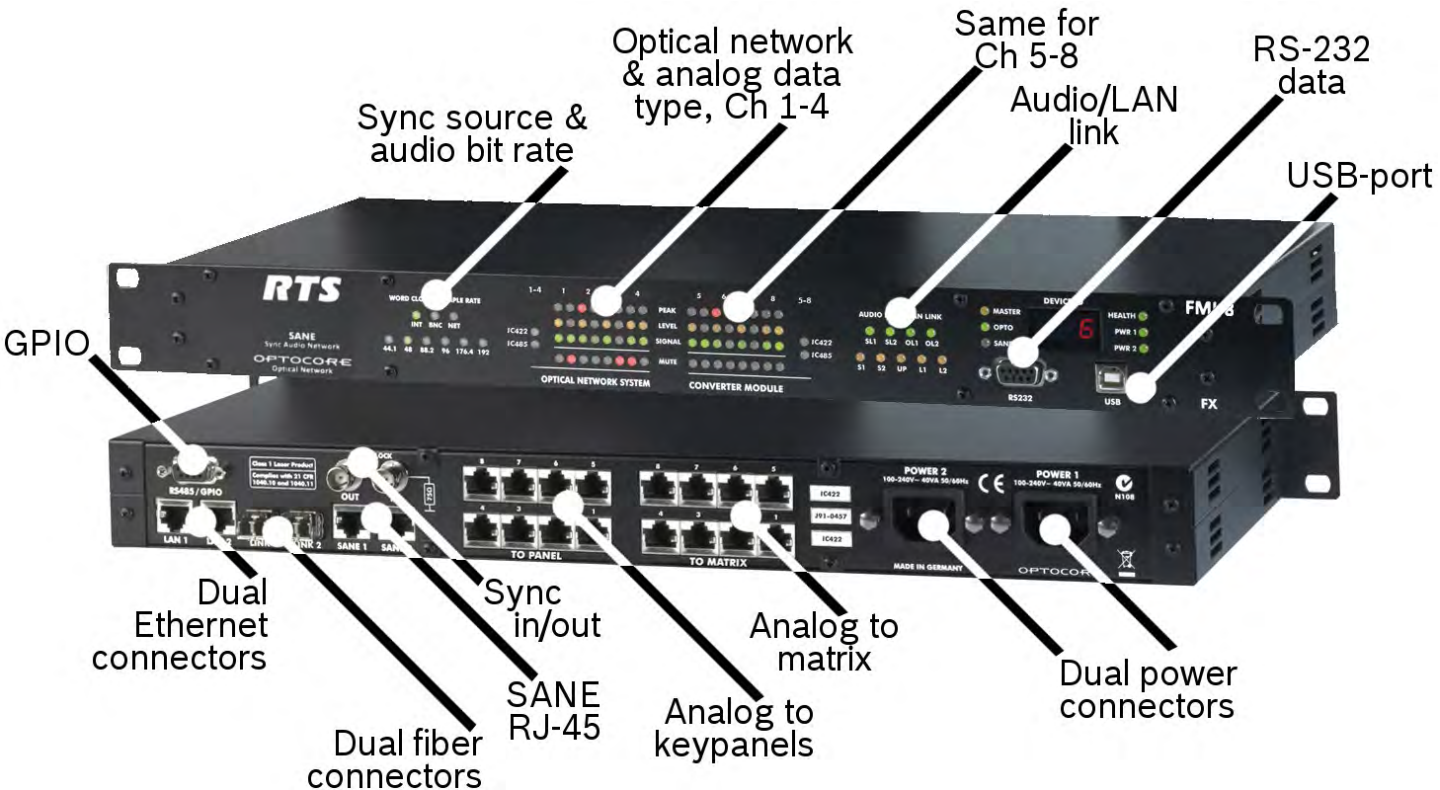
In the use case shown in the graphic below, a total of eight analog keypanels are connected to an ADAM-M, using a ring topology. Both multiplexers shown in the bottom of the graphic have four keypanels. Each of the green lines that make up the multiplexer ring actually consists of a pair of fibers, one for transmit and one for receive. Each multiplexer transmits on two fibers and receives on two fibers. If a fiber pair fails or is accidentally disconnected, only a few microseconds of audio are lost before the signal is re-established.

In addition to providing a fully redundant connection for each keypanel, the topology also simplifies the wiring. There is no need to pull an analog cable from each keypanel, all the way to the matrix. It is sufficient to connect it to the nearest multiplexer with an available port. Once a keypanel is connected to the multiplexer, it is possible to route it to an output port on any of the multiplexers in the ring. This means two matrices could even share the same optical ring.



PRODUCT VIEWS

FMI-8 front and rear



FMI-4 front



FMI-4 back



USE CASES

BACKGROUND

RTS is present in the aerospace industry. Aerospace industry has strong ties to national defense. The nature of the research is secret, and suppliers are often under NDAs (Non-Disclosure Agreements). For this reason, this Use Case does not divulge the identity of the company in question.

Activities that require real-time coordination to succeed are great applications for intercom. Keypanels are pre-programmed with all the individuals or groups with which each user communicates. By pressing a key, users establish an instantaneous bi-directional audio path. The matrix is the audio router that makes this possible. In aerospace, activities like testing new rocket engines or supervising an actual launch require coordination of hundreds of people. In the case of a launch, they may be in different geographical locations, so the ability to tie them together into an integrated network is critical to success.

In this particular Use Case, at least four locations communicate in real time during a launch:

- The launch site itself
- The company headquarters
- NASA’s Johnson Control Center
- The satellite manufacturer

A LAUNCH

At the launch sites, there are multiple operators that monitor subsystems of the rocket and supporting equipment. Computers, of course, monitor processes that may be too quick for a human to perceive. At a certain point in the countdown, computers trigger the events that have to happen in a certain sequence for the systems of the rocket to work correctly. At the company headquarters location, additional operators perform similar tasks. NASA’s Johnson Control Center is also tied into the communications system – they are the “flight control” of space and nobody is allowed to launch anything without involving them. Finally, the manufacturer of the payload – typically a satellite that needs to be delivered into orbit – monitors the health of the satellite itself (temperature, vibrations, electrical voltages and so on).



Figure 1. Space Shuttle, now decommissioned, was a symbol of space exploration

NETWORK DIAGRAM

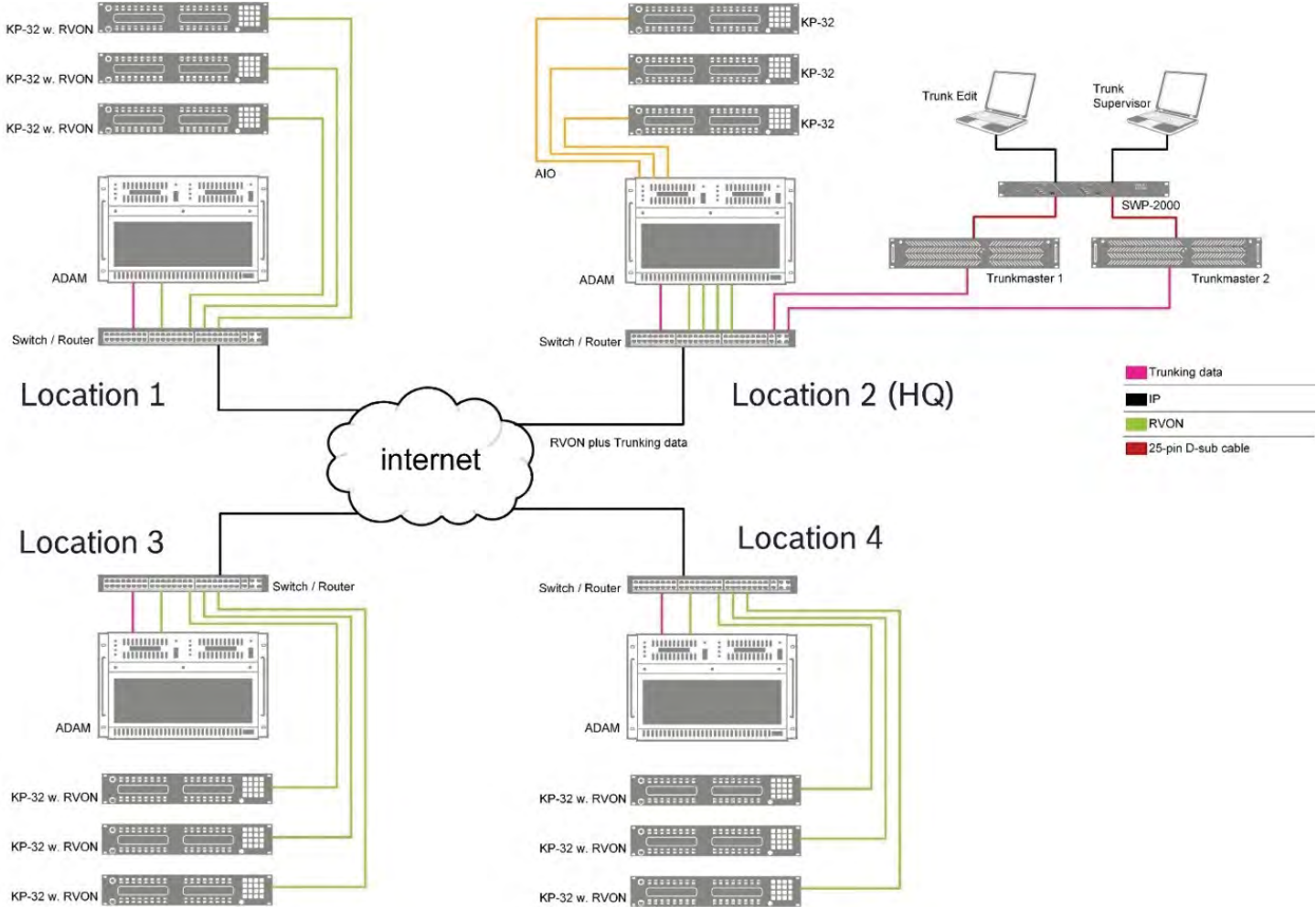


Figure 2. Four company locations are tied together with RVON technology

Figure 2 shows the four company locations. The two external locations are not shown (NASA and the satellite maker). However, they are also tied in with RVON. Each location has multiple keypanels. With the exception of one location, the company headquarters, keypanels with RVON capability are used. In the headquarters location, analog AIO is used. Some migration to OMNEO is occurring but this is not shown in the graphic. The OMNEO technology makes it easy to record multi-track audio directly to a server, including time stamping to allow the sequence of events to be re-created if necessary. An AZ-Edit function called enhanced tallies is used to indicate at the glance of an eye which facility is talking on a Partyline connection (not shown). Trunking is used for routing between matrices. The primary Trunkmaster is backed up by a redundant unit, and the switchover panel (SWP-2000) switches from primary to secondary if the primary unit goes down for any reason.

Note RVON is available as an optional and free firmware download on all KP-Series keypanels except the KP-3016A which is an analog-only version. In this example, KP-32 panels are shown. They require a plug-in card to use RVON.

BACKGROUND

Nuclear power plants use Uranium fuel to generate heat, which drives turbines to generate electricity. US nuclear plants shut down every 18 months to replace depleted fuel rods with fresh ones. During the refueling, technicians use wired and wireless communications equipment to coordinate their activities. Wireless gear is particularly important to high efficiency. There are at least two particular circumstances that make this application quite different from most other situations where intercom equipment gets used.

- During refueling, the reactor is shut down and revenue loss is on the order of one million dollars per day. Any investment in equipment that saves time is welcome.
- Equipment that has been used inside the so-called containment area is considered radioactive and can never come out again. Therefore, equipment can never be sent for service.

In addition, the use of multichannel audio through the containment wall saves having to compromise the building integrity.

ROLE OF RTS

Two decades ago, Nuclear Operating Companies (NOCs) used walkie-talkie equipment to communicate during refueling. Walkie-talkie is simplex only, meaning the channel can be used by one person, and everyone else needs to be listening, not broadcasting. That is why people say “over”: I am releasing the channel now. With walkie-talkies, refueling typically took between 60 and 90 days.

In the 1990s, NOCs started using UHF-based equipment from RTS. This product family is designated BTR followed by a number. BTRs allow full-duplex communication. Four beltpacks communicate to one base station, and base stations can be interconnected. This allows for much more integrated and efficient communications. Refueling time was cut to between 30 and 40 days. The equipment pays for itself! Two models are big-sellers with NOCs: The BTR-800 (shown in Figure 3) and the BTR-80N, a narrowband version of the BTR-800.



Figure 1. Cooling towers at a nuclear power plant



Figure 2. Two service technicians during a refueling operation. They can now communicate without shouting!



Figure 3. BTR-800 with four beltpacks

NETWORK DIAGRAM

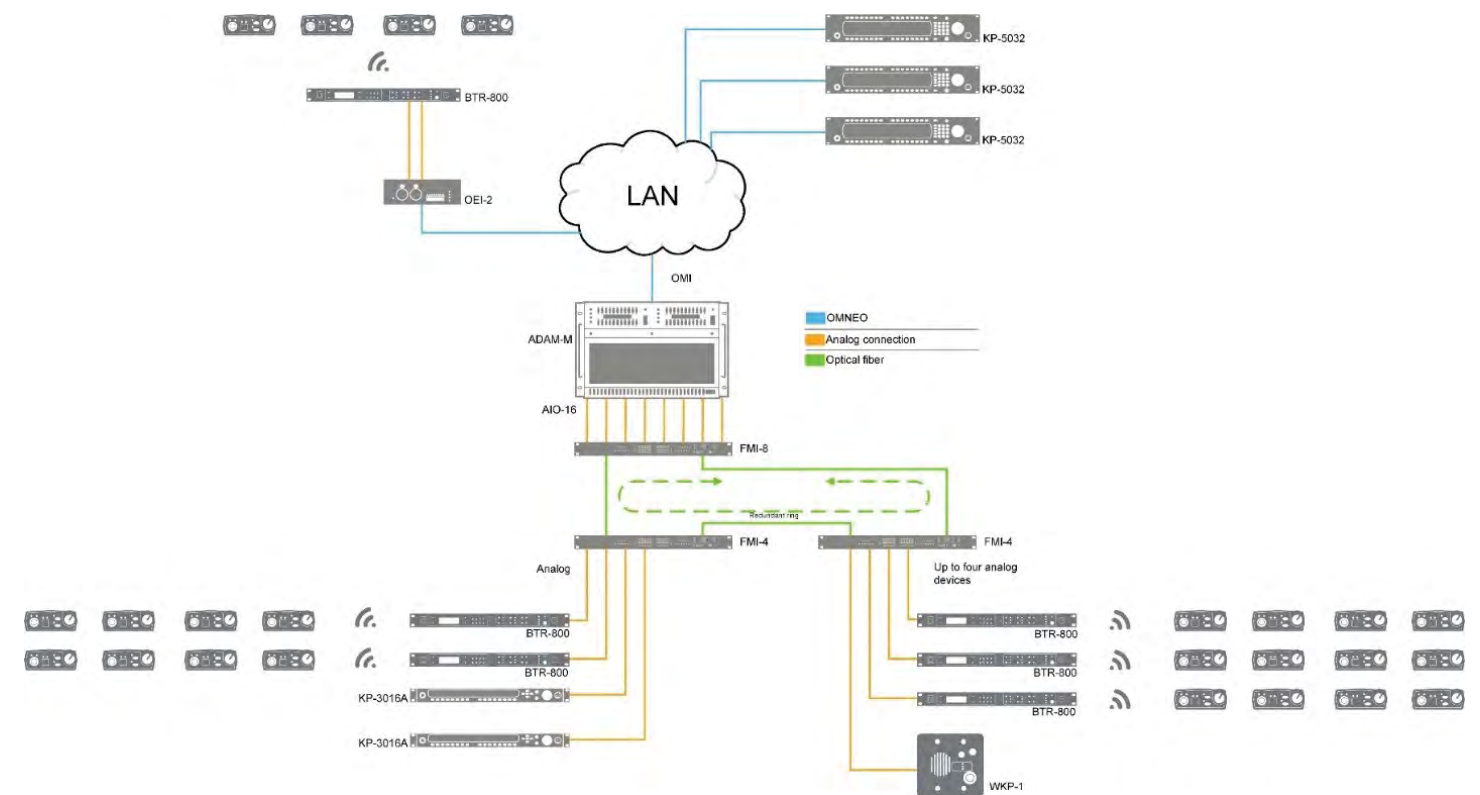


Figure 4. A matrix ties several BTR-800 together, as well as keypanels

In the diagram shown in Figure 4, and ADAM matrix is used to tie several network components together. In the upper left hand corner, a BTR-800 is connected via OMNEO. An OEI-2 converts the analog AIO format to the digital OMNEO format. Note the BTR has two analog lines. In the upper right hand corner, four keypanels are connected via OMNEO. Keypanels are used in the control room. In the lower portion, several BTRs are connected to the matrix via an optical ring. The FMI-Series (FMI-4 and FMI-8) fiber multiplexer interfaces are used to carry the analog signals over fiber, and to create redundancy. If any fiber in the green ring breaks, the audio is instantly rerouted over the alternate path. You can learn more about FMI-Series in a separate Application Note. Finally, a couple of different keypanels are shown, all connected over analog AIO. The KP-3016A is an analog-only keypanel in the KP-Series family. The WKP-1 is an analog one-channel keypanel developed for environmentally demanding applications.

BACKGROUND

Local television stations in the US are either Owned and Operated by major networks, such as ABC, CBS or NBC, or they are independently owned, but Affiliated by contract to a network. The networks supply local stations with content. Local television stations typically produce local news and weather.

ROLE OF RTS

RTS supplies the intercom system, typically a ODIN or a Cronus, as well as keypanels, partyline equipment and IFBs. Figure 1 shows an example of a local television station.

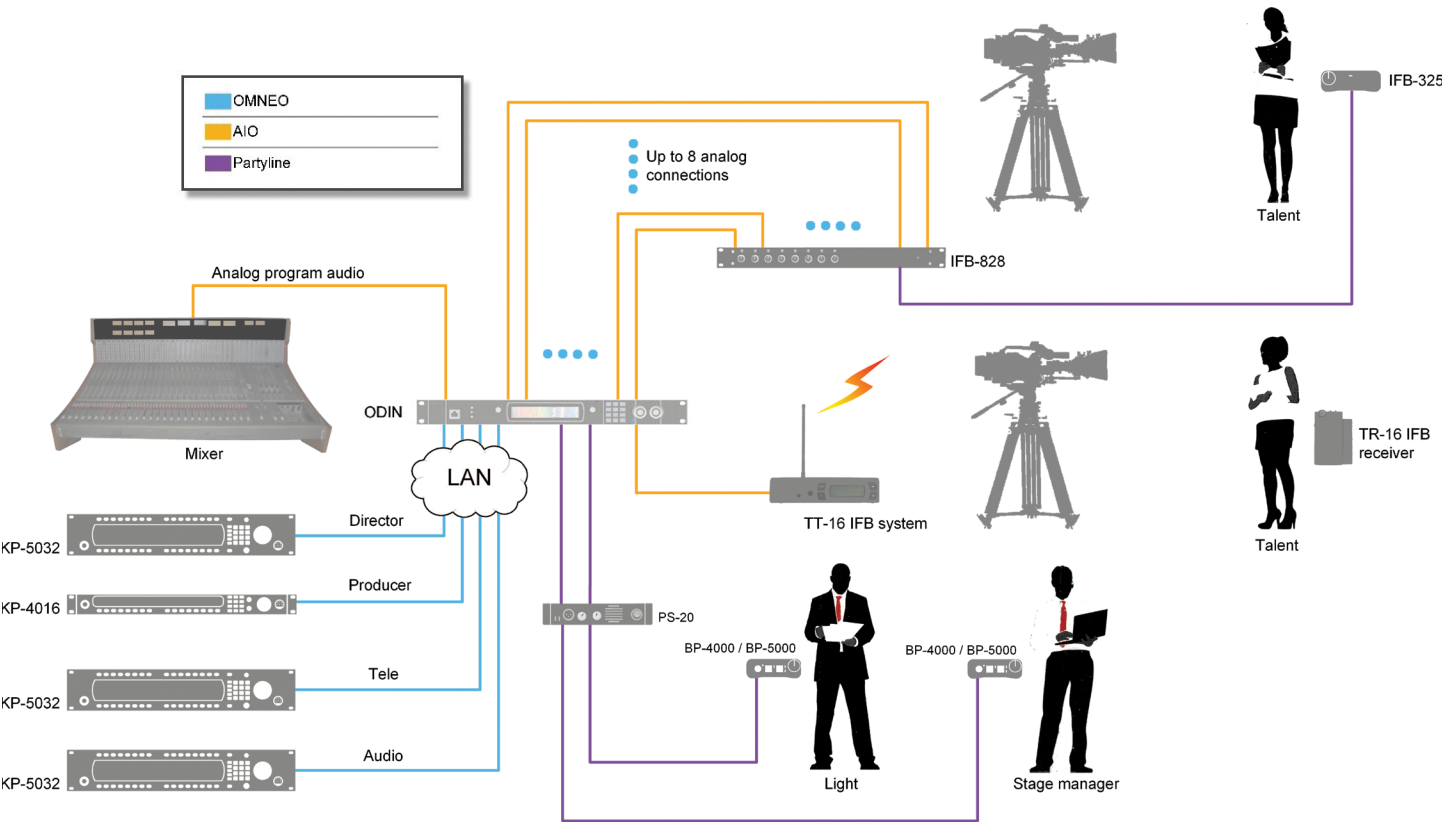


Figure 1. Small television station, overall view

DESCRIPTION

The ODIN at the center of Figure 1 is the central audio router. Analog program audio comes in from a mixer. Four keypanels are shown here for Director, Producer, Teleprompter, and Audio. A light technician and a stage manager each use wired partyline. In this example, there are two “talents” – one is using a wired IFB, the IFB-325, and one is using a wireless IFB, the TR-16. An IFB is a listen-only device. On-air talents typically have an ear-piece, in which they hear the program audio, also referred to as non-interrupt audio. However, when the on-air talent needs to hear an important message – the interrupt audio – the non-interrupt audio is dimmed or muted, to allow the important message to be heard. The message may be ‘going to commercial break in ten seconds’ or ‘wrap up your interview within 30 seconds’. Figure 2 shows how the IFB works.

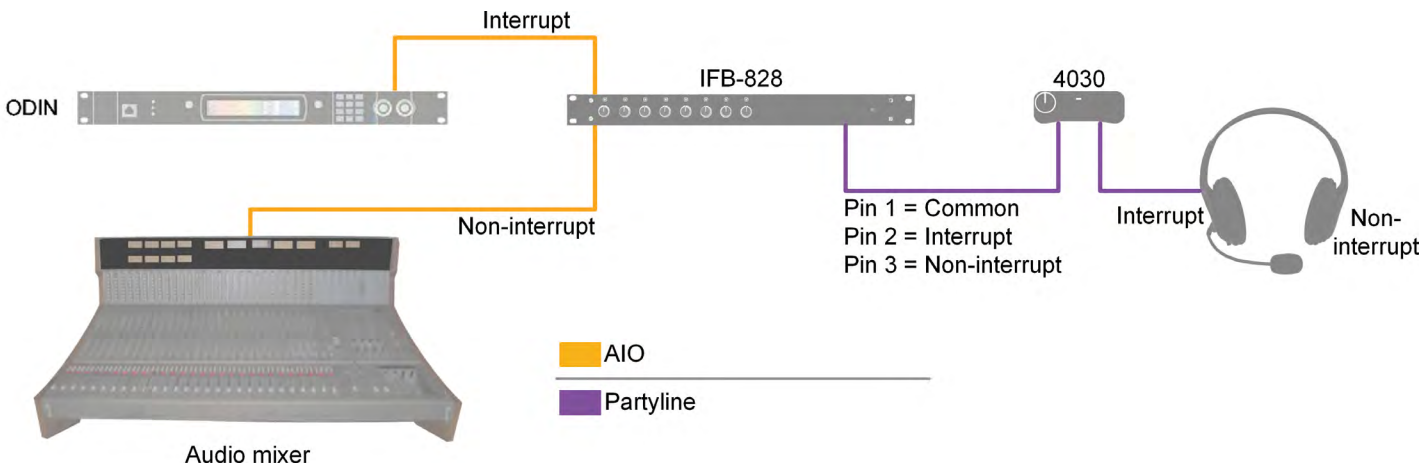


Figure 2. IFB - overview of principle of operation

In this example, non-interrupt audio comes from the audio mixer. The interrupt audio – which may be coming from the producer or director, is coming from the ODIN. The two audio sources converge in the IFB-828. In the RTS two-wire format, three wires are available. Interrupt and non-interrupt audio are transmitted on different pins. The IFB-325 is a passive device that allows the user to control the headset volume. Interrupt audio is fed to the headset on one ear, and the non-interrupt on the other. A wireless IFB system is also available. The transmitter is called TT-16 and the receiver is TR-16. The principle is the same but there is no wire to worry about. TT-16 / TR-16 operate in the VHF spectrum.

INTRODUCTION

Theaters often use a combination of partyline and matrix to coordinate light, stage technicians, stage managers, actors, as well as announcements to the audience, communications with the ticket booth etc.

A REAL EXAMPLE

Figure 1 shows a real-world example from a European theater located in Poland.

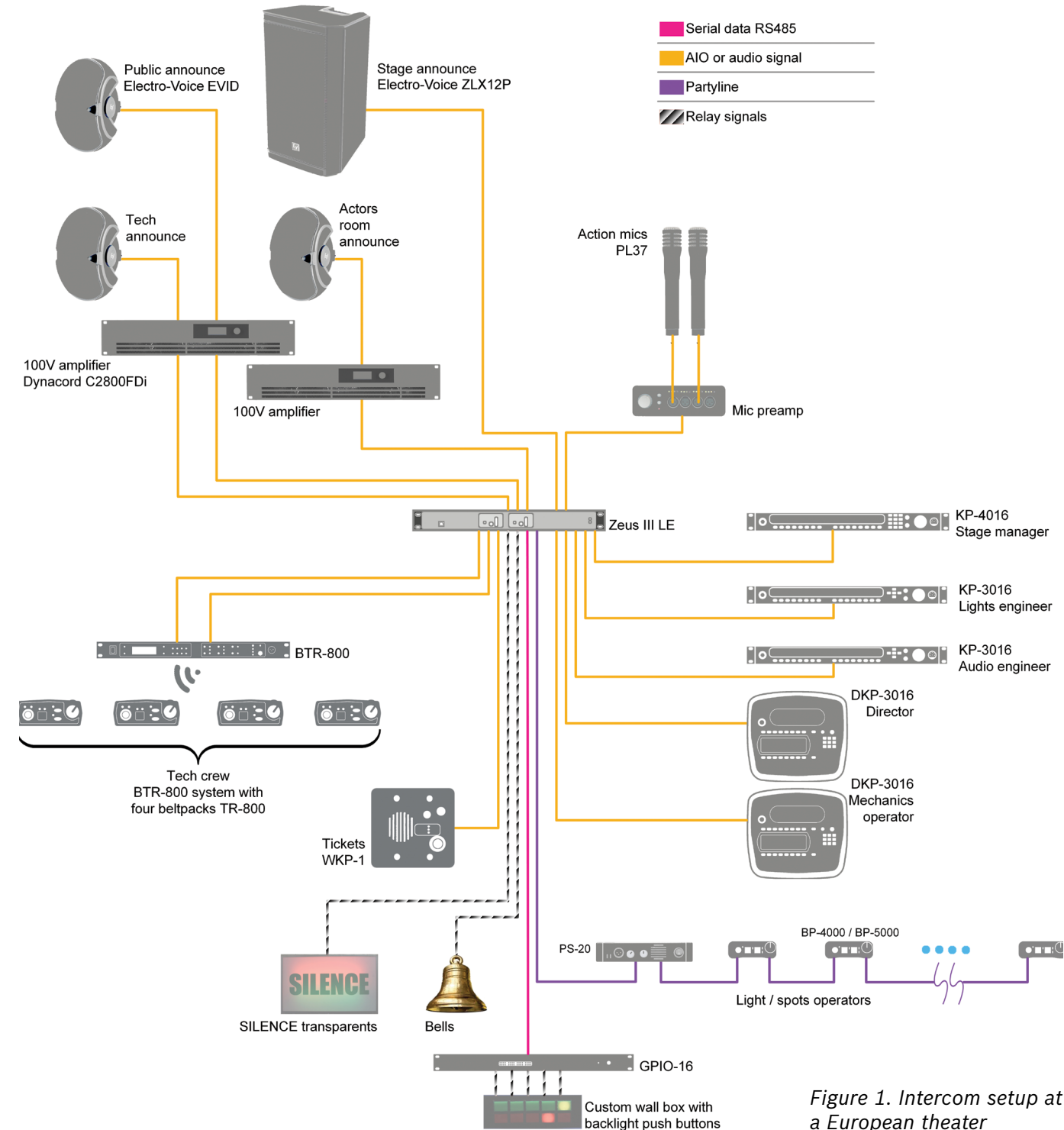


Figure 1. Intercom setup at a European theater

At the heart is a Zeus III LE, a 1RU matrix with 16 ports and support for two-wire analog partyline. Six keypanels are attached in this system: KP-Series keypanels are used for Stage Manager, Lights Engineer, Audio Engineer, Director, and Mechanics Operator. The ticket booth has a WKP-1, a ruggedized single-button keypanel that fits in a very compact form factor. A wireless partyline system, the BTR-800, is used for the Tech Crew. The system has two channels. The light and spotlight operators have wired beltacks. BP-4000 is a single-channel beltpack and the BP-5000 is a dual-channel beltpack. The power for the partyline is provided by a PS-20.

For audio announcements to the Public, Technician Crew, and Actors' room, this theater uses speakers from Electro-Voice, the EVID, which is powered from a 100V amplifier, in this case a Dynacord C2800FDi. For Stage announcements the ZLX12P is used. It has a built-in amplifier so it connects directly to the matrix. For action microphones, Electro-Voice PL37 is used. The signal goes through a microphone preamp before it goes into two of the matrix.

Finally, a separate GPIO-16 general purpose input/output device is used for a couple of customized functions. First, relay outputs from the GPIO-16 control the Silence transparents (lamps). Second, there is a custom wall box in the Actors' room. It has a set of backlit push buttons which can be used both to send indications, as well as allow actors to provide an acknowledgment (for example get ready to go on stage). The GPIO-16 communicates with the matrix through an RS-485 serial data connection. The keypanels are programmed to allow Director or Stage Manager to send signals to the Actors' room by actuating keys on their panel.



Figure 2. House of Culture in Poland - exterior view (top) and rehearsal of a theater production (bottom)

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