Development of Digitalized Narrow-band Mobile Radio

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Abstract

In recent years, because of insufficient wave frequency in mobile communications systems and stronger needs for multifunctionality through data transmission, demand has grown for the development of technologies that can improve the efficiency of radio-wave usage.

A major customer for our company's radios, the taxi industry, particularly the taxi industry in the Tokyo area where conventional analog systems have been used, has also been longing for the development of a digitalized narrow-band mobile radio. This is because authorization has not been given to use GPS-AVM systems (automatic vehicle-dispatch systems), which require separate data-dedicated frequencies.

Our company conducted research and development aimed at creating specifications for digitalized narrow-band mobile radio for taxis. Together with taxi industry groups, we conducted test runs aimed at achieving practical application and succeeded in developing this system as a standard. We were also the first company to develop a GPS-AVM digitalized narrow-band mobile radio that is suitable for use in taxis.

This report will introduce the technology that was developed in order to achieve specifications that are suitable for taxis. It will also provide a product overview of the digitalized narrow-band mobile radio that we recently developed.

1 Introduction

The popularization of mobile phone and other public mobile communication systems in recent years has been amazing. And even in the field of independent mobile communications systems, in which local governments and businesses have independently developed the systems, there have been increasing opportunities for digitalization as a means to effectively utilize radio waves and develop a wide variety of functions.

In 1998 the Telecommunications Technology Council held an inquiry received a report and proposed that the frequency bandwidth of each user be made narrower for more effective use. Receiving this proposal, the Ministry of Public Management, Home Affairs, Posts and Telecommunications (MPHPT, called the Ministry of Posts of Telecommunications at that time) revised the ministerial ordinance on the digitalization of narrow-band mobile radios in 1999, and a standard was created by the Association of Radio Industries and Businesses (ARIB).

In the field of independent mobile communications, our company has played a role in improving the efficiency of vehicle dispatching by taxi service providers by providing them with automatic vehicle monitoring (AVM) systems. One of these systems, the GPS-AVM system, which utilizes a global positioning system (GPS) to constantly monitor vehicle locations and achieve optimal dispatching of vehicles with respect to the locations of customers, has earned a good reputation among taxi service providers. However, because frequencies that are dedicated to data transmission are required when an analog-type radio is used, this system could not be introduced in the Tokyo area, which lacks sufficient frequency resources.

Thereupon, our company developed technology to achieve the performance required for a GPS-AVM system through digitalization of narrow-band mobile radios. Then using a radio that applied the developed technology, we performed successful test operations aimed at practical application. As a result, the addition of "a standard for GPS-AVM systems" to the previously established ARIB standard was proposed and recorded.

Hereinafter, an overview is provided in connection with the sale of digital radios for GPS-AVM systems.

2 GPS-AVM system

2.1 Overview of GPS-AVM system

Figure 1 shows an example of the system configuration. A summary of the system's operation will be explained hereinafter. Items ① through ⑧ correspond to the numbers in Figure 1.

①The positions of all vehicles in the system are monitored by GPS. When a vehicle moves a predetermined distance, information on the vehicle's position is sent via wireless transmission radio link to the vehicle positional information database in the vehicle dispatch center.

②When an order is received by telephone from a customer, the telephone number is sent to the host computer in the vehicle dispatch center via NTT's number display system.

③The host computer, which possesses a customer information database, uses the telephone number it receives to search for information on the customer.

④The customer information from ③ is compared to the vehicle positional information that was monitored in ①, and a search is made to find the vehicle that can move most quickly to the customer's location.

⑤Vehicle dispatch information for the searched vehicle (including the customer's address, name, and route) is sent to the base station.

⑥The vehicle dispatch information is sent to the designated vehicle from the base transceiver station via wireless transmission.

⑦After receiving dispatch instructions, the vehicle proceeds to pick the customer up.

⑧ Incidentally, atypical information, such an inquiry regarding belongings left behind, is communicated by voice via wireless transmission.

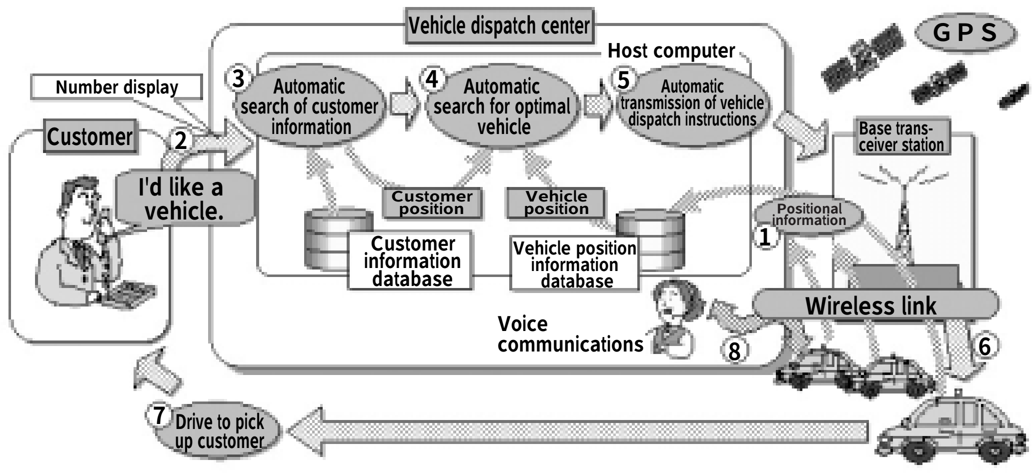


Fig.1 GPS-AVM system configuration (example)

2.2 Problems with development of digitalized narrow-band mobile radios

With a GPS-AVM system, three types of information, namely, ① the vehicle's positional information (vehicle→base), ⑥ dispatch instructional information (base→vehicle), and ⑧ voice information (vehicle←→base), as well as corresponding control signals, are sent using wireless transmission. However, when an analog (FM modulation)-type radio or conventional standard established by ARIB is used to construct this GPS-AVM system, problems occur, such as those described hereinafter.

(1) <u>Transmission efficiency is poor.</u> Since the quantity of vehicle positional information is small, the data length is short (burst transmission). But with the conventional standard, time was required to reproduce the transmission timing and improve transmission distortion for each burst transmission, so information could not be sent during this time. Furthermore, since the frequency of transmission is increased in order to improve the accuracy of vehicle position monitoring, the transmission efficiency further worsens, greatly affecting the system's overall operational efficiency.

(2) <u>The voice is interrupted</u> due to burst transmission and other data transmission. If controls are added to prevent this occurrence, data transmission will stop when voice appears and then be delivered as a group after the voice transmission ends. This leads to traffic congestion, which can prevent normal transmission.

(3) <u>The voice service area is small.</u> With a digitalized narrow-band mobile radio, voice is transmitted as data; thus, the service area is narrower than that of an analog system, making it necessary to set up a large number of base stations.

3 Principal technologies

This section will explain the technologies that were incorporated into the digitalized narrow-band mobile radio that was newly developed to solve these problems.

The achievements of this development work were examined by a subworking group whose membership included ten radio equipment manufacturers within ARIB, resulting in the establishment of ARIB STD-T61, Ver. 1.1, Appendix 1. [1)]

3.1 Small-quantity, highly frequent data transmission

Figure 2 shows a comparison of signal formats for digitalized narrow-band mobile radio's burst transmission. (a) shows the format for the standard that was established formerly by ARIB. To send 160-bit information, a 384-bit length is required (42% information occupancy rate). (b) shows a format that applies our company's developed technology. With it, 152-bit information can be transmitted with 192 bits, half the length of (a) (79% information occupancy rate).

This makes it possible to send positional data from the vehicle to the base at twice the frequency of transmission of a conventional digitalized narrow-band mobile radio, and four times the frequency of transmission of our company's analog-type system.

The conventional technology was improved in two ways:

(1) Reduction in bits for transmission timing reproduction
Elimination of P and reduction of S in Figure 2

(2) Elimination of transmission distortion compensation training signals
Elimination of LP in Figure 2

The improvements and contents are explained hereinafter.

3.1.1 Synchronous method

In Figure 2, P is a preamble signal used to determine the timing of the signal received (symbol timing). Normally, in a bit expression, a repetitive pattern of several bits, such as 1010 or 1001, is prescribed. The symbol S is a frame synchronizing signal that is used to determine the position (frame timing) of the basic unit (frame) of the information transmission. A designated bit pattern is prescribed for each system.

Table 1 is a comparison of (a) the conventional synchronization technique and (b) the newly developed synchronization technique. The conventional synchronization technique required two stages: the reproduction of symbol timing using a preamble signal, followed by the attainment of frame timing using a frame synchronizing signal. The newly developed synchronization method makes it possible to acquire symbol timing and frame timing at the same time by taking a comparison of the frame synchronizing signal and received signal , and then converting it

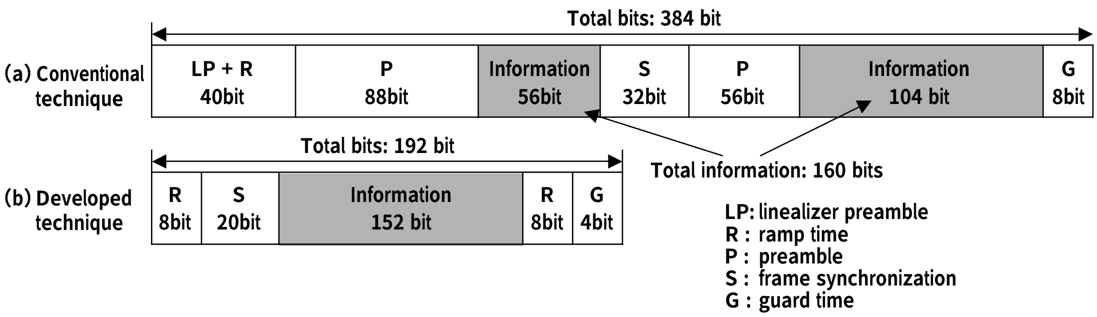


Fig.2 Burst transmission signal format

to an evaluation value.

The application of this synchronization technique has made the preamble signal unnecessary and made it possible to reduce the frame synchronizing signal from 32 bits to 20 bits.

Table 1 Comparison of synchronous acquisition methods

Item Compared	Conventional Technique	Newly Developed Synchronization Method
Synchronous stage	2 stages (symbol synchronization → frame synchronization)	1 stage (symbol synchronization + frame synchronization)
Synchronizing signal	2 types (preamble, frame synchronization)	1 type (frame synchronization)

3.1.2 High-linearity transmission power amplifier

In Figure 2, the LP in (a) is a training signal (linealizer preamble) that is used to compensate for the distortion of the high-frequency power amplifier (HPA). The R in (a) and (b) is the ramp time for controlling spectral spread by forming the signal rise and fall. Symbol G is the guard time for adjusting the radio wave's propagation delay time. It prevents signals from overlapping due to time differences when signals from multiple mobile stations arrive at the base station.

Also, the standard for ACPR (Note 1), which is established for the emission of digitalized narrow-band mobile radios, is strict because the bandwidth is narrow. High linearity is required for the HPA.

Figure 3(1) compares the configuration of (a) the conventional transmission technique and (b) the newly developed transmission technique. With the conventional transmission technique, distortion generated from insufficient HPA linearity is detected and fed back by the aforementioned LP and compensated by the digital signal processor (DSP). Since several tens of bits are consumed with this LP, this invites a drop in transmission efficiency.

With the newly developed technique, the HPA circuit's LD-MOS FET is given a push-pull configuration and negative feedback is provided. This enables the circuit to meet the ACPR standard. Figure 3(2) shows a comparison of characteristics.

The development of an HPA that has such high linearity made it possible to eliminate the LP and improve transmission efficiency. And by eliminating the detector circuit and reducing the compensation process, DSP specifications could be reduced and cost savings achieved.

3.2 Simultaneous transmission of voice and data

Next, simultaneous transmission of voice and data is an extremely effective way to solve the problem of voice being interrupted by data transmission, which was described in 2.2(2). With a digitalized narrow-band mobile radio, however, the transmission bandwidth per wave is narrow at approximately 5.8 kHz (channel spacing of 6.25 kHz). Considering the acquisition of the service area mentioned in 3.3, it is difficult to secure a transmission speed that exceeds 9600 bps.

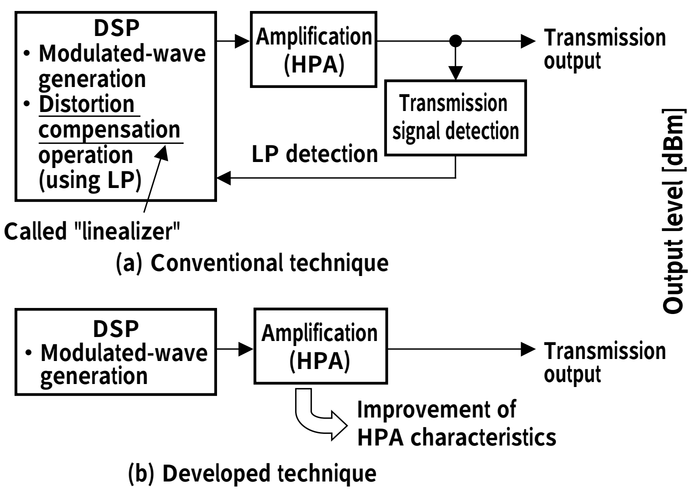
To achieve simultaneous transmission of voice and data under the aforementioned conditions, the following techniques were adopted:

(1) High-compressibility voice encoding

(2) Cyclical numbering of frames

Details are provided hereinafter. From this point, however, data other than voice will be described using the coinage "nonvoice"

(Note 1) ACPR: adjacent channel leakage power ratio. Refers to the ratio of the total emission power to the power leaked and radiated to an adjacent channel.



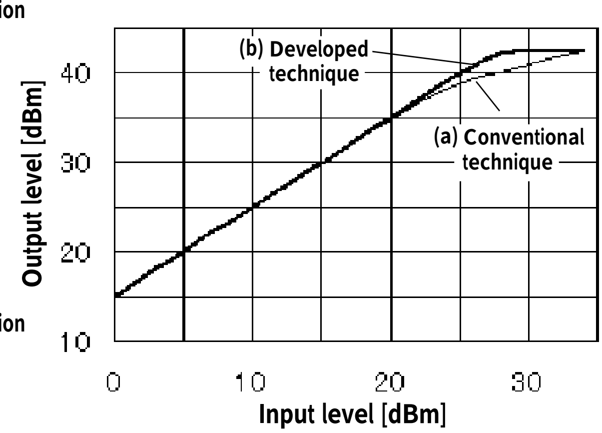


Fig.3 Comparisons of transmitter circuit configurations and input/output characteristics

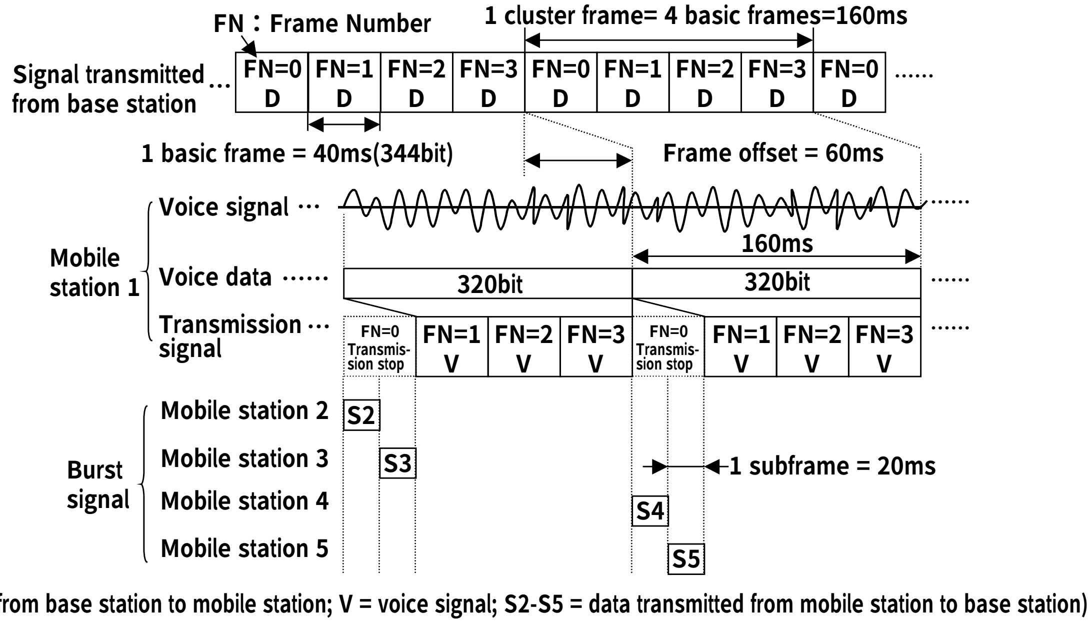


Fig.4 Frame configuration with superposed audio and data

3.2.1 Voice encoding (AMBE)

With a digitalized narrow-band mobile radio, voice signals are transmitted as voice data. On the receiving side, the voice data is converted to voice signals. Both voice quality and high compressibility are sought.

For the new system, Advanced Multi-Band Excitation (AMBE), developed by DVSI (U.S.), was adopted as the voice compression method (Note 2). AMBE was adopted not only because it can transmit voice as voice data at just 2000 bps but because the sound quality is close to natural.

3.2.2 Frame configuration

Figure 4 shows the configuration of a frame for achieving simultaneous transmission of voice and nonvoice. The length of one basic frame is 40 ms. Normally, the voice data of one basic frame is transmitted using one basic frame. With this development, however, the voice data of four basic frames is transmitted using three basic frames, while nonvoice data is superposed in the single empty basic frame.

If AMBE is adopted for voice CODEC, and voice signals are converted to 2000-bps voice data, the voice data of four basic frames (160 ms) will be 320 bits. If this is allocated to three basic frames, one basic frame becomes approximately 108 bits. Even if a strong error correction code is added to this, it can sufficiently handle the quantity transmitted with one basic frame (344 bits). Thus, it is possible to use three basic frames to send the voice of four basic frames.

Then there is the problem of determining the rule by which the voice data is distributed to each basic frame and then restored at the receiving end. For this, four basic frames are defined as a "cluster frame,", and the numbers (frame numbers: FNs) 0, 1, 2, and 3 are cyclically allocated to each basic frame within the cluster frame. Before a basic frame is transmitted, the FN and voice/nonvoice classification bit are added.

Moreover, it is established that a mobile station will transmit at a frame timing that is delayed by 60 ms (frame offset) from the frame timing transmitted by the base station, and voice will be transmitted with FN = 1, 2, 3. Using the FN and voice/nonvoice classification bit, the receiving side can accurately restore the original voice and nonvoice.

In addition, a 20-ms-length "subframe" was defined for the burst signal sent from the mobile station to the base station. Even if other mobile stations transmit voice, subframes can be used to transmit two station's data in 160 ms from a mobile station to the base station.

In this way it has become possible not only to send from a base station to a mobile station but to send mixed voice and nonvoice from different mobile stations to the base station.

(Note 2) This is referred to asvoice CODEC (abbreviation of Coding/DECording). It is a method of mutually converting voice signals and compressed voice data.

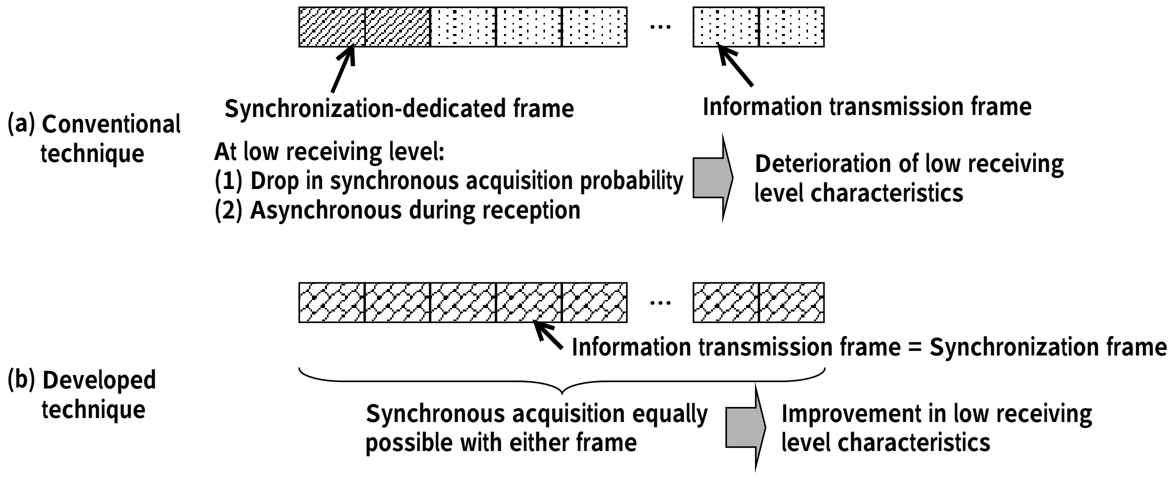


Fig.5 Improvement of synchronous acquisition characteristics with low reception voltage

3.3 Securing of voice service area

With mobile communications it is not unusual for the channel quality to reach a bit error rate (BER) of 10%, making channel quality incomparably harsher than that of wired communications. Since acute deterioration of the bit error rate is a characteristic of mobile communication channels, the voice service area of a digital system is generally narrower than that of an analog system, and a large number of base stations must be established.

To broaden the service area as much as possible, SCPC (single channel per carrier) was adopted as the access method for this development project, because it has advantages in securing receiver sensitivity and can secure a broad service area for a single base station. And although the transmission speed of the modulation method is slower than that of other methods, a (/4 shift QPSK system was adopted because of its strong fading strength and equipment cost advantages.

Moreover, as a result of the following two technical improvements, the system secures a voice service area that is equivalent to that of an analog system:

(1) Improved error correction performance
(2) Improved synchronous acquisition characteristics

3.3.1 Improved voice at a error correction performance

With the conventional technique, data to which an error correction had been added would be about twice as long as the original data to be transmitted. By adopting AMBE for the voice CODEC and improving the error correction performance by allocating the reduced portion of the transmitted voice data to the error correction code, it became possible to secure a service area. Convolutional code was used for the error correction code.

3.3.2 Improved synchronous acquisition characteristics

Figure 5 is a comparison of the frame configuration of (a) the conventional technique and (b) the newly developed technique.

The format of Figure 2(a) is used for the conventional technique's synchronization-dedicated frame. Although a portion of the information is superposed as mentioned in 3.1, most portions are used for synchronous acquisition. A different format has been provided for the information transmission frame that is used to send information.

If, with the conventional technique, the received level drops due to fading and synchronous acquisition cannot be attained during reception of the synchronization-dedicated frame in Figure 5(a), synchronous acquisition is achieved with the information transmission frame. However, the received level required for synchronous acquisition will be higher than when an information transmission frame is synchronized with a synchronous acquisition-dedicated frame. Consequently, if synchronous acquisition fails initially, receiver characteristics will naturally worsen when the received level is low.

The high-speed synchronization method mentioned in 3.1.1 was applied during this development project. When high-speed synchronization is used, all of the frames are information transmission frames and can be made to function as synchronization frames. As shown in Figure 5(b),receiver characteristics improve because synchronization can immediately be acquired when the received level recovers, regardless of the portion of received level that drops during a series of transmissions.

3.4 Test operation

To verify whether the digital and analog service areas are equivalent, data was acquired during April to June, 2001, from drive tests performed over the same course in Tokyo and Saitama Prefecture, using a recently developed digitalized narrow-band mobile radio's test model and a conventional analog-type radio. As a result it was confirmed that the service area was equivalent to that of an analog-type during voice transmission. [2) 3)]

Then using a prototype that was provided with a data transmission function and improved voice service quality, drive tests were performed in Tokyo in March 2003. With the focus on data transmission, the tests compared the digitalized narrow-band mobile radio prototype to our company's existing system and confirmed the prototype's practical application over a broader area than our company's analog system.

4 Summary of digitalized narrow-band mobile radio

The newly developed digitalized narrow-band mobile radio consists of an analog processing unit, which is made up of transmitting and receiving circuits, and a digital processing unit, which performs modulation/demodulation and control.

The main specifications of the radio are shown in Table 2, while the radio's features are set forth and its configuration shown in Figure 6.

Table 2 Specifications of digitalized narrow-band mobile radio

№	Item	Specifications
1	Transmission/reception frequency	450～470MHz
2	Access method	SCPC
3	Communication method	Simplex
4	Modulation system	π/4 shift QPSK
5	Type of emission	5K80G1D,5K80G1E
6	Total transmission speed	9600bps
7	Voice encoding method	AMBE(2000bps)
8	Antenna power	2W(average, mobile station)
9	Maximum bandwidth	3MHz(reception sensitivity: 3dB decrease)
10	Antenna impedance	50Ω
11	Channels	10
12	Channel spacing	6.25kHz
13	Oscillation method	Synthesizer method with crystal oscillation control
14	Reception method	Double super heterodyne
15	Transmission method	Double conversion
16	Power supply	DC13.8V(negative ground)
17	External dimensions	W150×H45×D140mm

4.1 Frequency configuration

The digitalized narrow-band mobile radio utilizes orthogonal modulation. To achieve orthogonal modulation/demodulation with an analog device, know-how and labor are required to eliminate carrier leaks. In the hardware configuration, too, two systems are required for the DA converter, AD converter, and filter. For this reason, the cost can be high.

Thereupon, the newly developed radio achieves orthogonal modulation/demodulation by means of a digital signal processor (DSP). As for the frequency configuration, it would be ideal, from a cost perspective, to use a direct conversion system that directly converts high-frequency signals and pre-orthogonal-modulation signals with a DSP. In this project, however, the configuration was adopted such that high-frequency waves are converted to intermediate-frequency waves during processing. This was done in order to achieve desired values for the interference characteristics, in order to adjacent frequencies in transmission performance, and to achieve desired values for the selective characteristics from adjacent frequency signals during receiving performance.

4.2 Analog processing unit

To convert intermediate-frequency signals and 400-MHz-band high-frequency signals, four local oscillators, two for sending and two for receiving, are normally used. By generating these four oscillations based on a single standard oscillator, the newly developed system achieves compactness and lower cost.

Also, the transmitting unit takes the intermediate-frequency signals that are generated by the DSP and converts the frequency twice to create 400-MHz-band high-frequency signals. After 400-MHz-band high-frequency signals are created, they are amplified to the prescribed electric power by the aforementioned HPA. The receiving unit then receives the 400-MHz-band high-frequency signals and converts the frequency twice to create intermediate-frequency signals, and inputs them in the DSP.

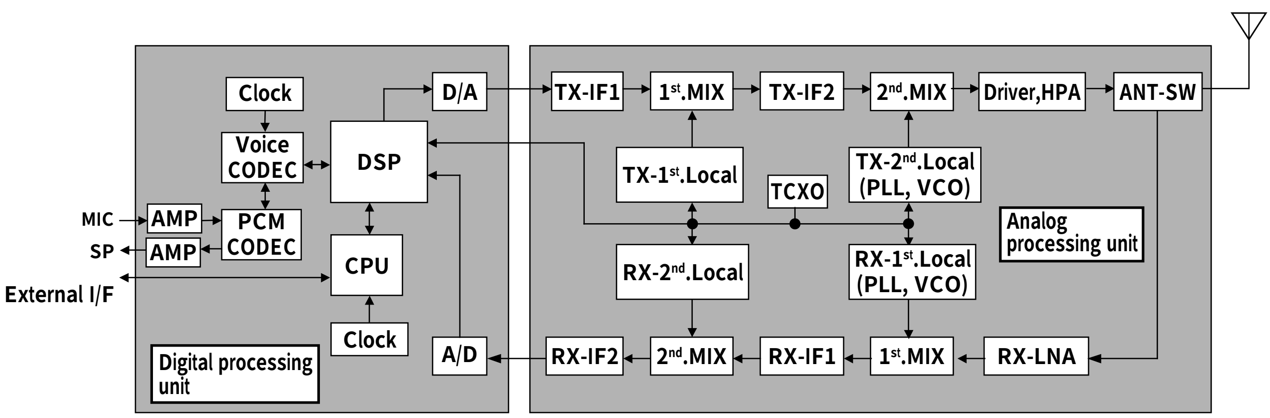


Fig.6 Configuration of digitalized narrow-band mobile radio

5 Conclusion

As previously described, we developed a digitalized narrow-band mobile radio that is optimal for taxi GPS-AVM systems. This enables us to provide taxi service providers with a system that is more efficient than conventional analog systems.

We hope to continue to develop products that will contribute to the improvement of our customers' operations by constantly pursuing further cost reduction, compactness, and higher performance. We also hope to apply the results of this development work to areas outside the taxi business and will make tireless efforts aimed at the development of our society.

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