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Millimeter-Wave Propagation Through a Terrestrial Atmosphere

GEORGE HUFFORD

Because at millimeter wavelengths the atmosphere is not only inhomogeneous but also dispersive, rays are bent by amounts that depend on frequency. For a normal, well-mixed atmosphere, the resulting frequency distortion can be estimated by a simple technique involving a complex-valued effective earth's radius.

At millimeter-wave frequencies, the terrestrial atmosphere is absorptive, dispersive, and inhomogeneous. Because it is inhomogeneous, rays are bend by refraction, and because it is dispersive, the amount of bending depends on the frequency. Thus the path length between two terminals depends on frequency, and this means there is an additional frequency distortion to be accounted for. In this letter, we shall derive an "effective refractivity" to do such an accounting. We shall also show that the correction involved is rather small.

Restricting ourselves to the two-dimensional scalar problem, we begin with a polar coordinate system having its origin at the center of the earth. We suppose a source at $(r_0, 0)$ and an observation point at (r, θ) and we define

$$R'^2 = r'^2 + r_0'^2 - 2r'r_0' \cos \theta' \tag{1}$$

where

$$r' = \frac{a}{\gamma}\left(\frac{r}{a}\right)^{\gamma} \qquad r_0' = \frac{a}{\gamma}\left(\frac{r_0}{a}\right)^{\gamma} \tag{2}$$

$$\theta' = \gamma\theta \tag{3}$$

and where a is the radius of the earth and γ will be determined shortly. It then follows that the function

$$\phi(r, \theta) = H_0^{(1)}(kn_1R') \tag{4}$$

satisfies the two-dimensional Helmholtz equation

$$\nabla^2\phi + k^2n^2\phi = 0 \tag{5}$$

where k and n_1 are constants and

$$n = n_1\left(\frac{r}{a}\right)^{\gamma - 1}. \tag{6}$$

The transformation from the unprimed to the primed coordinates in (2) and (3) is a conformal transformation which may be pictured as a change to an "effective earth's radius" equal to a/γ. Note, for example, that the great circle distance between source and observation point is

$$x = a\theta = (a/\gamma)\theta' \tag{7}$$

and that if h, h_0 are heights above the earth we have the approximations

$$r' \approx (a/\gamma) + h \qquad r_0' \approx (a/\gamma) + h_0 \tag{8}$$

and

$$n \approx n_1\left(1 + \frac{\gamma - 1}{a} h\right). \tag{9}$$

The idea, then, is to suppose that n represents the index of refraction of the terrestrial atmosphere and so to set n_1 equal to its value at $h = 0$ and γ so that

$$\gamma = 1 + \frac{a}{n}\frac{dn}{dh}\bigg|_{h=0}. \tag{10}$$

This scheme works quite well at microwave frequencies where for a normal, well-mixed atmosphere the formulas (6) and (9) are an *excellent representation* [1] *of how the refractive index varies with* height for the first kilometer or so, and γ is often measured to have the convenient value of 3/4. It seems clear that the scheme should also work for millimeter waves. Because of atmospheric absorption, the derivative in (10), and therefore the value of γ, is complex. This ruins the simple geometry of the transformed variable, but it has no effect on the arithmetic that leads to (5).

We thus have an exact solution to a problem that is a good imitation to how millimeter waves should propagate through a normal atmosphere. To complete the picture, we need values for n and for the gradient dn/dh. We set

$$n(f, h) = 1 + N_0(h) + N_1(f, h) \tag{11}$$

where $N_0 + N_1$ is the refractivity of the atmosphere, N_0 being the nondispersive part and N_1 the dispersive part. The value of N_0 is real, positive, and independent of the frequency f, while N_1 is complex and a nontrivial function of f.

Using Liebe's model [2] of atmospheric characteristics, we have drawn in Fig. 1 the frequency dependence of the real and imaginary parts of the dispersive part of the refractivity. Conditions assumed were that of the ICAO standard atmosphere at sea level and a 50 percent relative humidity. The value of N_0 under these same conditions is 310 N units.

In Fig. 2 we have drawn, versus frequency, the corresponding lapse rate of refractivity. This is the total lapse rate and includes the nondispersive part. Note that the particular 50 percent value of relative humidity makes the lapse rate at zero frequency equal to about 39 N units/km. This is what it must be if γ is to have the conventional value of 3/4.

To estimate what effect the numbers in Figs. 1 and 2 would imply, we must find a suitable approximation to the function in (4). For this, we first note that when source and observation points are separated by more than a few wavelengths, we may replace the Hankel function by its asymptotic expansion: the exponential exp (ikn_1R') multiplied by a slowly varying amplitude that can be ignored.

To estimate R' it is convenient, and not at all restrictive, to suppose that a is slightly changed to lie midway between r_0 and r. Then setting

$$h = y/2 \qquad h_0 = -y/2 \tag{12}$$

we expand the cosine term in (1) and extend the expressions in (8) to terms of third order. We find

$$R'^2 \approx x^2 + y^2 + \frac{(\gamma - 1)(\gamma - 2)}{12a^2} y^4 - \frac{\gamma}{4a^2} x^2y^2 - \frac{\gamma^2}{12a^2} x^4. \tag{13}$$

If we further assume that the path is mostly horizontal so that y/x is of the same order of smallness as x/a, we have

$$R' \approx R_0'\left(1 - \frac{\gamma_0}{12}\frac{x^2}{a}\frac{\partial N_1}{\partial h}\right) \tag{14}$$

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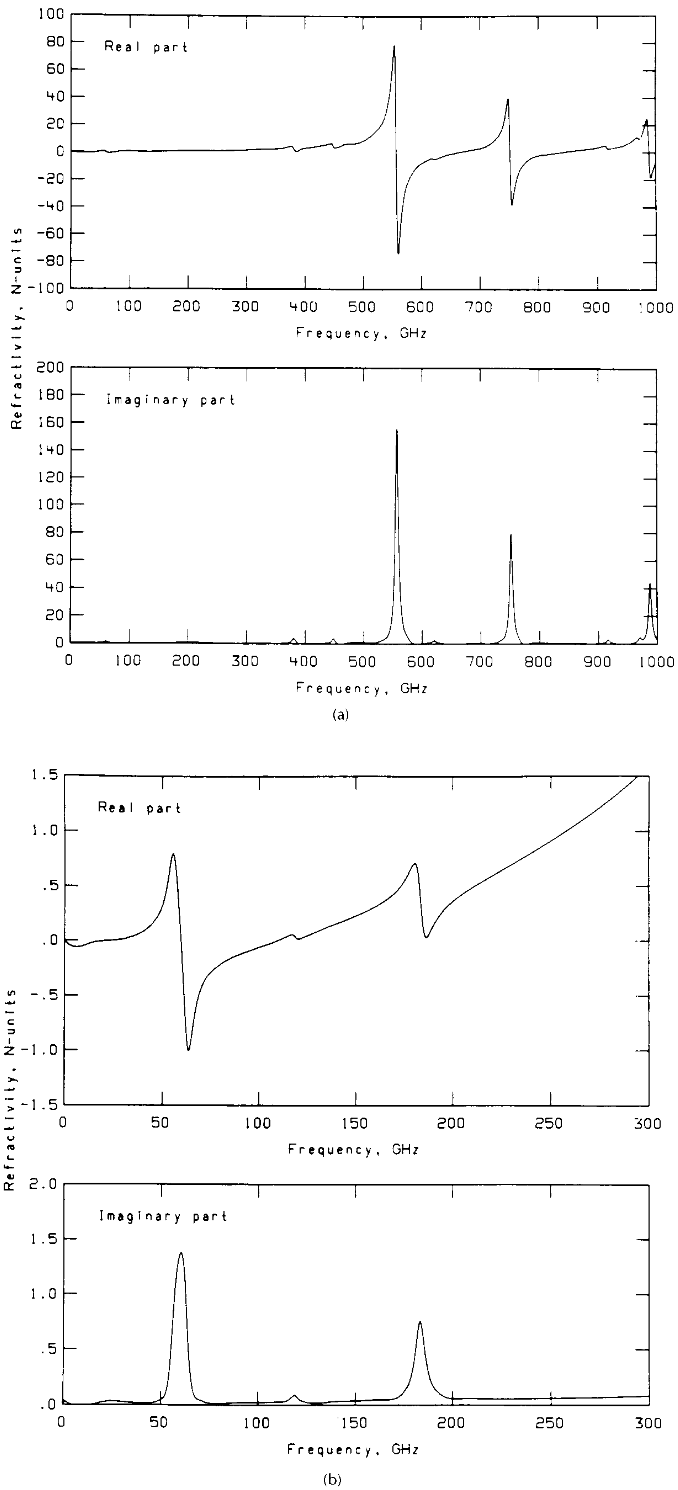

Fig. 1. (a) The dispersive part of the refractivity of the atmosphere. The ICAO standard atmosphere at sea level is assumed with 50-percent relative humidity. (b) An expanded version of Fig. 1(a).

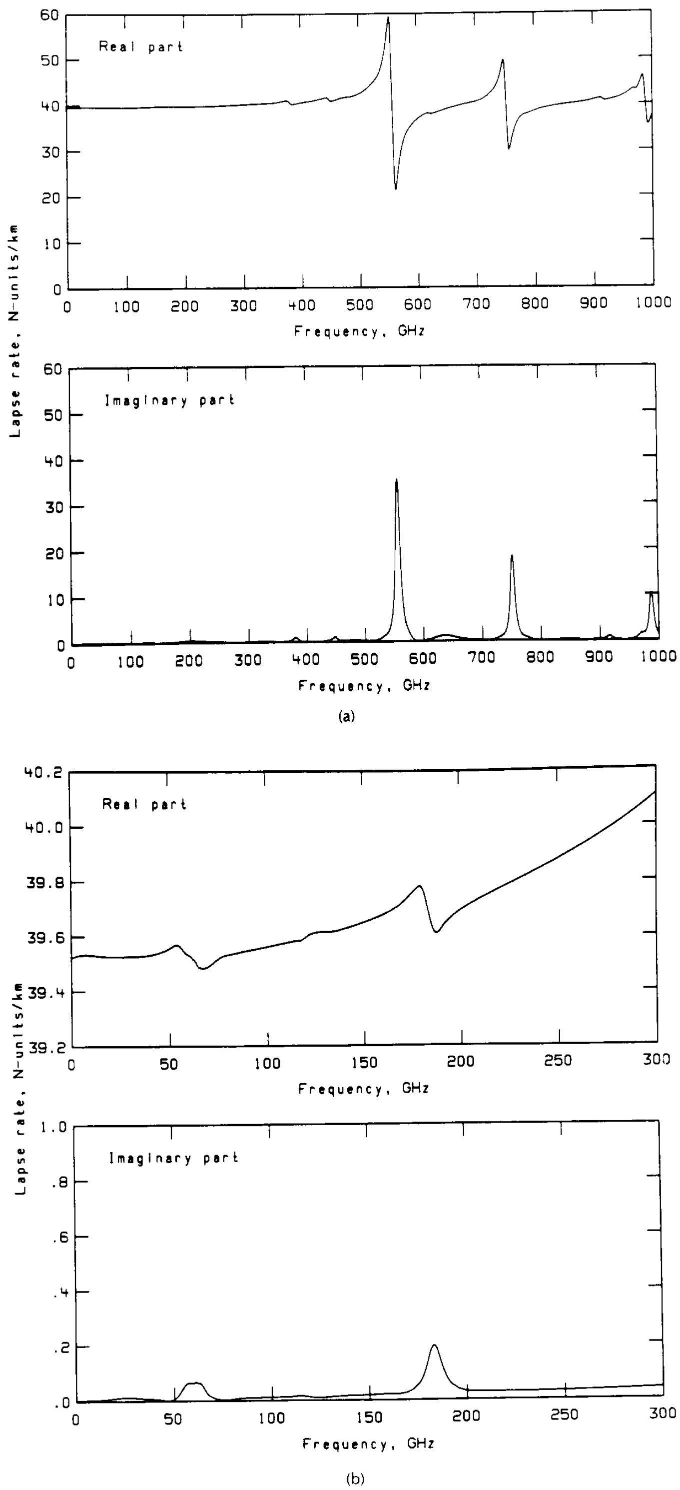

Fig. 2. (a) The lapse rate of atmospheric refractivity for the standard atmosphere at sea level. (b) An expanded version of Fig. 2(a).

where R_0' and γ_0 are the values of R' and γ at zero frequency. Finally, we combine (11) and (14) to obtain

$$kn_1R' \approx k\left(1 + N_0 + N_1 - \frac{\gamma_0}{12}\frac{x^2}{a}\frac{\partial N_1}{\partial h}\right)R_0'. \quad (15)$$

In this way, we have transformed the problem of a varying atmosphere over a spherical earth into a simple, frequency-dependent perturbation of the atmospheric refractivity. If x is as large as 50 km, the coefficient in front of the gradient here becomes equal to 0.025 km. This should be multiplied by the values in Fig. 2 and then compared with those in Fig. 1. For frequencies less than 300 GHz, we would be adding to the refractivities something on the order of 0.02 or 0.03 N units. Even near the enormous line at 557 GHz, the extra term, which might be as large as 1 N unit, would be dwarfed by the values of refractivity that are already there.

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An Efficient Multiplexing Technique for Packet-Switched Voice/Data Networks

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For synchronization of packetized voice in a packet-switched voice/data network, we propose an efficient multiplexing scheme which uses a synchronous frame format. This scheme has discrete delay characteristics, and provides a simple play-out method for reproduction of voice signal. We investigate its performance by obtaining the cumulative distribution of delay of voice packets and the mean waiting time of data packets.

I. Introduction

Presently, the packet switching approach is known to be an effective method for integrating voice and data in a single link. In a packet-switched network, voice packets are normally transmitted synchronously to reconstruct voice signal with acceptable quality. Various reconstruction algorithms using time stamps have been suggested for voice synchronization [1], [2].

One may note that the synchronization procedure of voice packets required for reconstruction of voice *at the receiver* is basically a "passive" method, and its implementation is not simple. So far, this approach has mainly been considered [1], [2]. On the contrary, synchronous transmission of packet streams *in the network* may be viewed as an "active" approach, and is desirable from the implementation point of view. For the synchronization of voice packets in the network, the network delay should have a fixed duration. But, when a multiple of voice sources are integrated on a single link, one cannot have the network delay of a fixed duration only by the priority service of voice because of the asynchronous nature of the packet-switched network. Therefore, for synchronous transmission in the packet-switched network it would be necessary to utilize a master frame format of time division multiplexing (TDM).

In the packetization process of voice signals, packets are generated periodically, while voice sources are active [3]. To synchronize voice packets inside the network, we utilize the sequence numbers and the inter-arrival time of the received voice packets. In order to estimate the packet generation interval and active periods of voice sources from the received packet streams, we use a frame structure which has the same duration as a voice packet generation interval (VPGI). In this case, synchronization of voice packets can be done without additional packet headers such as time stamps. With the proposed frame structure, each voice packet would experience network transmission delay of one VPGI or its multiple.

II. System Description

The system being studied is depicted in Fig. 1. In this figure, N voice sources generate packets with fixed length every T seconds while active, and each packet stream is synchronously served at the start of each frame time T based on the first-come first-served rule. The trunk capacity is equal to the output capacity of M voice channels. This means that M voice packets can be served during a frame time T in the trunk. We assume that there are two separate buffers for voice and data, and that voice is served with priority over data. The overflown packets in a frame duration are stored in the buffer for service in the next frame. Therefore, the stored packets in the buffer require additional waiting time of an integer multiple of a frame duration (or a VPGI). These discrete delay characteristics of voice packets make it possible to reproduce the voice signal without processing the time information of voice packets.

As for data packets, they are transmitted during the remaining frame time after voice packets have been served. When the remaining frame duration is short for a data packet, it is stored for service in the next frame. As a result, the frame utilization may be degraded for the price of synchronization of voice packets.

III. Analysis of Transmission Delay

To investigate the proposed frame structure, let the buffer sizes of voice and data packets at the start of the nth frame be q_n^v and q_n^d, respectively.[1] Then, the following relationships hold:

$$q_{n+1}^v = (q_n^v - M)^+ + a_n^v \quad (1)$$

$$q_{n+1}^d = (q_n^d - v_n^d)^+ + a_n^d \quad (2)$$

where a_n^v and a_n^d denote the numbers of voice and data packets arrived at the system during the nth frame duration, respectively; M and v_n^d are the maximum numbers of voice and data packets being served during the nth frame duration, respectively; and the superscript + denotes a nonnegative value.

After some algebraic manipulation using (1) and (2), one can show that the moment generating functions (mgf's) of the buffer sizes for voice and data packets, $Q^v(z)$ and $Q^d(z)$, are represented in the steady state, respectively, by

$$Q^v(z) = \frac{\sum_{k=0}^{M-1} P[q_n^v = k] - z^{-M}\sum_{k=0}^{M-1} P[q_n^v = k]z^k}{1 - z^{-M}G_a^v(z)}\, G_a^v(z) \quad (3)$$

$$Q^d(z) = \frac{\sum_{k=0}^{\infty} P[q_n^d = k]\left(1 - \sum_{l=0}^{k} P[v_n^d = l]\right) - \sum_{k=0}^{\infty}\sum_{l=0}^{k-1} P[q_n^d = l]P[v_n^d = k]z^{-k+l}}{1 - V^d(z^{-1})G_a^d(z)}\, G_a^d(z) \quad (4)$$

where $G_a^v(z)$, $G_a^d(z)$, and $V^d(z)$ are the mgf's of the random variables a_n^v, a_n^d, and v_n^d, respectively, and $P[\cdot]$ denotes the probability density function. The queueing statistics given by (3) and (4) may be obtained, once the source models of voice and data packets are determined. In (3), $G_a^v(z)$ can be obtained using a Markov chain model for active voice sources [3]. Assuming that the queueing behavior of voice packets is stationary and that data sources are

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[1] We use the superscripts "v" and "d" for voice and data, respectively.

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