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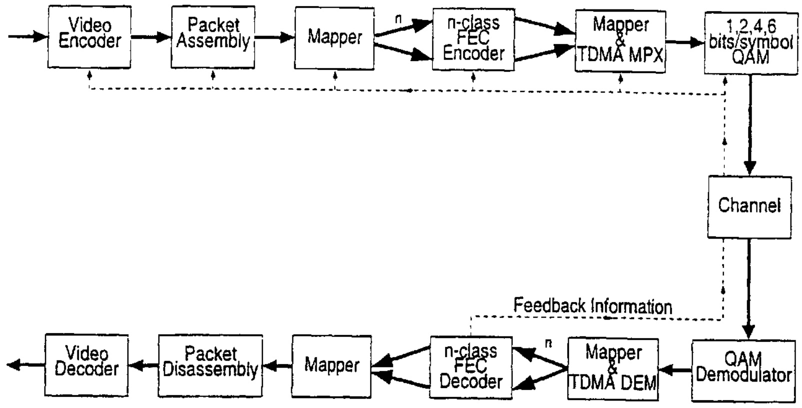

(57) **Abstract:** A range of Adaptive Orthogonal Frequency Division Multiplex (AOFDM) video systems are proposed for interactive communications over wireless channels. The proposed constant target bitrate subband adaptive OFDM (CTBR-AOFDM) modems can provide a lower BER, than a corresponding conventional OFDM modem. The slightly more complex switched time-variant target bitrate adaptive OFDM TVTBR-AOFDM modems can provide a balanced video quality performance, across a wider range of channel SNRs. The main advantage of the proposed technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, etc. Furthermore, when the mobile is roaming in a hostile outdoor propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

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Title of the Invention

ADAPTATIVE OFDM TRANSMITTER

1 Background of the Invention

The invention relates to adaptive Orthogonal Frequency Division Multiplexing (OFDM) based transmission of multimedia signals, such as interactive video or audio, speech etc.

In contrast to the *burst-by-burst reconfigurable wideband* multimedia transceivers described in this document, the term *statically reconfigurable* found in this context in the literature refers to multimedia transceivers that cannot be near-instantaneously reconfigured. More explicitly, the previously proposed *statically reconfigurable* video transceivers were reconfigured on a long-term basis under the base station's control, invoking for example in the central cell region - where benign channel conditions prevail - a less robust, but high-throughput modulation mode, such as 4 bit/symbol Quadrature Amplitude Modulation (16QAM), which was capable of transmitting a quadruple number of bits and hence ensured a better video quality. By contrast, a robust, but low-throughput modulation mode, such as 1 bit/symbol Binary Phase Shift Keying (BPSK) can be employed near the edge of the propagation cell, where hostile propagation conditions prevail. This prevented a premature hand-over at the cost of a reduced video quality.

The philosophy of the fixed, but programable-rate proprietary video codecs and statically reconfigurable multi-mode video transceivers presented by Streit *et al.* for example in References [1] was that irrespective of the video motion activity experienced, the specially designed video codecs generated a constant number of bits per video frame. For example, for videophony over the second-generation Global System of Mobile Communications known as the GSM system at 13 kbps and assuming a video scanning rate of 10 frames/s, 1300 bits per video frame have to be generated. Specifically, two families of video codecs were designed, one refraining from using error-sensitive run-length coding techniques and exhibiting the highest possible error resilience and another, aiming for the highest possible compression ratio. This fixed-rate approach had the advantage of requiring no adaptive feedback controlled bitrate fluctuation smoothing buffering and hence exhibited no objectionable video latency or delay. Furthermore, these video codecs were amenable to video telephony over fixed-rate second-generation mobile radio systems, such as the GSM.

The fixed bitrate of the above proprietary video codecs is in contrast to existing standard video codecs, such as the Motion Pictures Expert Group codecs known as MPEG1 and MPEG2 or the ITU's H.263

codec, where the time-variant video motion activity and the variable-length coding techniques employed result in a time-variant bitrate fluctuation and a near-constant perceptual video quality. This time-variant bitrate fluctuation can be mitigated by employing adaptive feed-back controlled buffering, which potentially increases the latency or delay of the codec and hence it is often objectionable for example in interactive videophony. The schemes presented by Streit *et al.* in References [1] result in slightly variable video quality at a constant bitrate, while refraining from employing buffering, which again, would result in latency in interactive videophony. A range of techniques, which can be invoked, in order to render the family of variable-length coded, highly bandwidth-efficient, but potentially error-sensitive class of standard video codecs, such as the H.263 arrangement, amenable to error-resilient, low-latency interactive wireless multimode videophony was summarised in [2]. The adaptive video rate control and packetisation algorithm of [2] generates the required number of bits for the burst-by-burst adaptive transceiver, depending the on the capacity of the current packet, as determined by the current modem mode. Further error-resilient H.263-based schemes were contrived for example by Färber, Steinbach and Girod at Erlangen University [3], while Sadka, Eryurtlu and Kondoz [4] from Surrey University proposed a range of improvements to the H.263 scheme. Following the above portrayal of the prior art in both video compression and statically reconfigurable narroband modulation, let us now consider the philosophy of wideband burst-by-burst adaptive quadrature amplitude modulation (AQAM) in more depth.

In burst-by-burst adaptive modulation a higher-order modulation scheme is invoked, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, a more robust lower order modulation scheme is employed, when the channel exhibits inferior channel quality, in order to improve the mean Bit Error Ratio (BER) performance. A practical scenario, where adaptive modulation can be applied is, when a reliable, low-delay feedback path is created between the transmitter and receiver, for example by superimposing the estimated channel quality perceived by the receiver on the reverse-direction messages of a duplex interactive channel. The transmitter then adjusts its modem mode according to this perceived channel quality.

Recent developments in adaptive modulation over a narrow-band channel environment have been pioneered by Webb and Steele [5], where the modulation adaptation was utilized in a Digital European Cordless Telephone - like (DECT) system. The concept of variable rate adaptive modulation was also advanced by Sampei *et al* [6], showing promising advantages, when compared to fixed modulation in terms of spectral efficiency, BER performance and robustness against channel delay spread. In another paper, the numerical upper bound performance of adaptive modulation in a slow Rayleigh flat-fading channel was evaluated by Torrance *et al*[7] and subsequently, the optimization of the switching threshold levels using Powell minimization was used in order to achieve a targeted performance [8, 9]. In addition, adaptive modulation was also studied in conjunction with channel coding and power control techniques by Matsuoka *et al* [6] as well as Goldsmith *et al.*[10].

In the narrow-band channel environment, the quality of the channel was determined by the short term Signal to Noise Ratio (SNR) of the received burst, which was then used as a criterion in order to choose the appropriate modulation mode for the transmitter, based on a list of switching threshold levels, l_n [5, 9]. However, in a wideband environment, this criterion is not an accurate measure for judging the quality of the channel, where the existence of multi-path components produces not only power attenuation of the transmission burst, but also intersymbol interference. Subsequently, a new criterion has to be defined to estimate the wideband channel quality in order to choose the appropriate modulation scheme.

2 Summary of the Invention

Particular and preferred aspects of the invention are set out in the accompanying independent and dependent claims. Features of the dependent claims may be combined with those of the independent claims as appropriate and used in combinations other than those explicitly set out in the claims.

The performance benefits of OFDM symbol-by-symbol adaptive modulation are described, employing a higher-order modulation mode on those OFDM subcarriers, where the frequency-domain channel transfer function is favourable, ie does not exhibit a high attenuation, or at subchannel frequencies, where the signal is unimpaired by co-channel interferers. This procedure is employed, in order to increase the system's bits per symbol (BPS) capacity and conversely, invoking a more robust, lower order modulation mode, when the channel exhibits inferior channel quality.

Two specific embodiments are described, a fixed bitrate and a time-variant bitrate system. The fixed-rate system allocates a fixed number of bits to each OFDM symbol, mapping the bits on to the highest-quality subcarriers. Hence this system optimises the bit allocation across the frequency domain, but ignores the time-variant nature of the channel quality. Therefore the associated bit error rate (BER) will be time-variant. By contrast, the time-variant bitrate system adjusts the number of bits mapped to the OFDM symbol on a time-variant basis, depending on the instantaneous channel quality. Hence it endeavours to optimise the bit allocation versus both time and frequency. This bit allocation policy allows us to maintain a near-constant BER versus time.

It is shown that due to the described adaptive modem mode switching regime a seamless multimedia source-signal representation quality - such as video or audio quality - versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advantage of the described technique is - in particular in the context of the time-variant bitrate embodiment - that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This can achieved on a near-instantaneous or OFDM symbol-by-symbol adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when a mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

3 Brief Description of the Drawings

For a better understanding of the invention and to show how the same may be carried into effect reference is now made by way of example to the accompanying drawings, in which:

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4 Detailed Description

4.1 State-of-the-art

Burst-by-burst adaptive quadrature amplitude modulation (AQAM) was contrived by Steele and Webb [5], in order for the transceiver to cope with the time-variant channel quality of narrowband fading channels. Further related research was conducted at the University of Osaka by Sampei and his colleagues, investigating variable coding rate concatenated coded schemes [6], at the University of Stanford by Goldsmith and her team, studying the effects of variable-rate, variable-power arrangements [10] and at Southampton University in the UK, investigating a variety of practical aspects of AQAM [11, 12]. The channel's quality is estimated on a burst-by-burst basis and the most appropriate modulation mode is selected in order to maintain the required target bit error rate (BER) performance, whilst maximizing the system's Bit Per Symbol (BPS) throughput. Using this reconfiguration regime the distribution of channel errors becomes typically less bursty, than in conjunction with non-adaptive modems, which potentially increases the channel coding gains. Furthermore, the soft-decision channel codec metrics can be also invoked in estimating the instantaneous channel quality, irrespective of the type of channel impairments.

A range of coded AQAM schemes were analysed by Matsuoka *et al* [6], Lau *et al* [13] and Goldsmith *et al* [10]. For data transmission systems, which do not necessarily require a low transmission delay, variable–throughput adaptive schemes can be devised, which operate efficiently in conjunction with powerful error correction codecs, such as long block length turbo codes. However, the acceptable turbo interleaving delay is rather low in the context of low-delay interactive speech. Video communications systems typically require a higher bitrate than speech systems and hence they can afford a higher interleaving delay.

The above principles - which were typically investigated in the context of narrowband modems - were further advanced in conjunction with wideband modems, employing powerful block turbo coded wideband Decision Feedback Equaliser (DFE) assisted AQAM transceivers [14]. A neural-network Radial Basis Function (RBF) DFE based AQAM modem design was proposed in [15], where the RBF DFE provided the channel quality estimates for the modem mode switching regime. This modem was capable of removing the residual BER of conventional DFEs, when linearly non-separable received phasor constellations were encountered.

The above burst-by-burst adaptive principles can also be extended to Adaptive Orthogonal Frequency Division Multiplexing (AOFDM) schemes [16] and to adaptive joint-detection based Code Division Multiple Access (JD-ACDMA) arrangements [17]. The associated AQAM principles were invoked in the context of parallel AOFDM modems also by Czylwik et al [18], Fischer [19] and Chow et al [20].

Adaptive subcarrier selection has been advocated also by Rohling et al [21] in order to achieve BER performance improvements. Due to lack of space without completeness, further significant advances over benign, slowly varying dispersive Gaussian fixed links - rather than over hostile wireless links - are due to Chow, Cioffi and Bingham [20] from the USA, rendering OFDM the dominant solution for asymmetric digital subscriber loop (ADSL) applications, potentially up to bitrates of 54 Mbps. In Europe OFDM has been favoured for both Digital Audio Broadcasting (DAB) and Digital Video Broadcasting [22, 23] (DVB) as well as for high-rate Wireless Asynchronous Transfer Mode (WATM) systems and for the new HIPERLAN standard due to its ability to combat the effects of highly dispersive channels. The idea of 'water-filling' - as allocating different modem modes to different subcarriers was referred to - was proposed for OFDM by Kalet [24] and later further advanced by Chow et al [20]. This approached was rendered later time-variant for duplex wireless links for example in [16]. Lastly, the co-channel interference sensitivity of OFDM can be mitigated with the aid of adaptive beam-forming in multi-user scenarios.

Our main contribution is that upon invoking the technique advocated - irrespective of the channel conditions experienced - the transceiver achieves always the best possible video quality by automatically adjusting the achievable bitrate and the associated video quality in order to match the channel quality experienced. This is achieved on a near-instantaneous basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile outdoor propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

4.2 AOFDM Signalling Scenarios

AOFDM transmission parameter adaptation is an action of the transmitter in response to time–varying channel conditions. It is only suitable for duplex communication between two stations, since the transmission parameter adaptation relies on some form of channel estimation and signalling. In order to efficiently react to the changes in channel quality, the following steps have to be taken:

- *Channel quality estimation:* In order to appropriately select the transmission parameters to be employed for the next transmission, a reliable prediction of the channel quality during the next active transmit timeslot is necessary.

- *Choice of the appropriate parameters for the next transmission:* Based on the prediction of the expected channel conditions during the next timeslot, the transmitter has to select the appropriate modulation schemes for the subcarriers.

- *Signalling or blind detection of the employed parameters:* The receiver has to be informed, as to which set of demodulator parameters to employ for the received packet. This information can either be conveyed within the packet, at the cost of loss of useful data bandwidth, or the receiver can attempt to estimate the parameters employed at the transmitter by means of blind detection mechanisms.

Depending on the channel characteristics, these operations can be performed at either of the duplex stations, as shown in Figures 1(a), 1(b) and 1(c). If the channel is reciprocal, then the channel quality estimation for each link can be extracted from the reverse link, and we refer to this regime as open–loop adaptation. In this case, the transmitter needs to communicate the transmission parameter set to the receiver (Figure 1(a)), or the receiver can attempt blind detection of the transmission parameters employed (Figure 1(c)).

If the channel is not reciprocal, then the channel quality estimation has to be performed at the receiver of the link. In this case, the channel quality measure or the set of requested transmission parameters is communicated to the transmitter in the reverse link (Figure 1(b)). This mode is referred to as closed–loop adaptation.

4.3 Video Transceiver

The schematic of the whole system is depicted in Figure 2. The multimedia source signal generated by the video encoder of Figure 2 is assembled into transmission packets constituting an OFDM symbol and the bits may be additionally mapped by the Mapper of Figure 2 to n number of different Forward Error Correction (FEC) protection classes. These bits are then convenveyed to the optional Time Division Multiplex (TDMA) scheme of Figure 2, before they are assigned to the OFDM subcarriers of the adaptive QAM modem seen in Figure 2.

As a particular embodiment of the proposed system concept, in this study we investigate the transmission of 704x576 pixel Four-times Common Intermediate Format (4CIF) high-resolution video sequences at 30 frames/s using a subband-adaptive turbo-coded OFDM transceiver. The transceiver can modulate 1, 2 or 4 bits onto each OFDM sub-carrier, or simply disable transmissions for sub-carriers which exhibit a high attenuation, or phase distortion due to channel effects. We note, however that the proposed principles are applicable to arbitrary multimedia source signals, bit rates, source signal representation quality, or even different channel codecs.

The main advantage of the proposed technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

The H.263 video codec exhibits an impressive compression ratio, although this is achieved at the cost of a high vulnerability to transmission errors, since a run-length coded bitstream is rendered undecodable by a single bit error. In order to mitigate this problem, when the channel codec protecting the video stream is overwhelmed by the transmission errors, we refrain from decoding the corrupted video packet, in order to prevent error propagation through the reconstructed video frame buffer [2]. We found that it was more beneficial in video quality terms, if these corrupted video packets were dropped and the reconstructed frame buffer was not updated, until the next video packet replenishing the specific video frame area was received. The associated video performance degradation was found perceptually unobjectionable for packet dropping- or transmission frame error rates (FER) below about 5%. These packet dropping events were signalled to the remote video decoder by superimposing a strongly protected one-bit packet acknowledgement flag on the reverse-direction packet, as outlined in [2]. Turbo error correction codes were used. The associated parameters will be discussed in more depth during our further discourse.

4.4 Comparing subband-adaptive to fixed modulation mode transceivers

In order to show the benefits of the proposed subband-adaptive OFDM transceiver, we compare its performance to that of a fixed modulation mode transceiver under identical propogation conditions, while having the same transmission bitrate. The subband-adaptive modem is capable of achieving a low bit error ratio, since it can disable transmissions over low quality sub-carriers and compensate for the lost throughput by invoking a higher modulation mode, than that of the fixed-mode transceiver over the high-quality sub-carriers.

Table 1 shows the system parameters for the fixed BPSK and QPSK transceivers, as well as for the corresponding AOFDM transceivers. The system employs constraint length 3, 1/2-rate turbo coding, using octal generator polynomials of 5 and 7, and random interleavers. Hence the unprotected bitrate is about half the channel coded bitrate. The protected to unprotected bitrate ratio is not exactly half, since two tailing bits are required to reset the convolutional encoders' memory to their default state in each

	BPSK mode	QPSK mode
Packet rate	4687.5 Packets/s	
FFT length	512	
OFDM Symbols/Packet	3	
OFDM Symbol Duration	2.6667μs	
OFDM Time Frame	80 Timeslots = 213μs	
Normalised Doppler frequency, f'_d	1.235×10^{-4}	
OFDM symbol normalised Doppler frequency, F_D	7.41×10^{-2}	
FEC Coded Bits/Packet	1536	3072
FEC-coded video bitrate	7.2Mbps	14.4Mbps
Unprotected Bits/Packet	766	1534
Unprotected bitrate	3.6Mbps	7.2Mbps
Error detection CRC (bits)	16	16
Feedback error flag bits	9	9
Packet header Bits/Packet	11	12
Effective video Bits/Packet	730	1497
Effective video bitrate	3.4Mbps	7.0Mbps

Table 1: System parameters for the fixed QPSK and BPSK transceivers, as well as for the corresponding subband-adaptive OFDM (AOFDM) transceivers for Wireless Local Area Networks (WLANs).

transmission burst. In both modes a 16-bit CRC is used for error detection, and 9 bits are used to encode by simple repetition coding the reverse link feedback acknowledgement information. The feedback flag decoding ensues using majority logic decoding. The packetisation requires a small amount of header information added to each transmitted packet, which is 11 and 12 bits/packet for BPSK and QPSK, respectively. The effective video bitrates for the BPSK and QPSK modes are then 3.4 and 7.0 Mbps.

The fixed mode BPSK and QPSK transceivers are limited to one and two bits per symbol, respectively. However, the AOFDM transceivers operate at the same bitrate, as their corresponding fixed modem mode counterparts, although they can vary their modulation mode on a sub-carrier by sub-carrier basis between 0, 1, 2 and 4 bits per symbol. Zero bits per symbol implies that transmissions are disabled for the sub-carrier concerned.

The "micro-adaptive" nature of the subband-adaptive modem is characterised by Figure 3, portraying at the top a contour plot of the channel SNR for each subcarrier versus time. At the centre and bottom of the figure the modulation mode chosen for each 32-subcarrier subband is shown versus time for the 3.4 and 7.0 Mbps subband-adaptive modems, respectively. The channel SNR is also shown in a three-dimensional form in Figure 4, which maybe more convenient to visualise. It can be seen that when the channel is of high quality – like for example at about frame 1080 – the subband-adaptive modem used the same modulation mode, as the equilivalent fixed rate modem in all subcarriers. When the channel is hostile – like around frame 1060 – the subband-adaptive modem used a lower-order modulation mode in some subbands, than the equivalent fixed mode, or in extreme cases disabled transmission for that subband. In order to compensate for the loss of throughput in this subband a higher-order modulation mode was used in the higher quality subbands.

One video packet is transmitted per OFDM symbol, therefore the video packet loss ratio is the same, as the OFDM transmission frame error ratio. The video packet loss ratio is plotted versus the channel SNR in Figure 6. It is shown in the graph that the subband-adaptive transceivers – or synonymously termed as microscopic-adaptive (μAOFDM), in contrast to OFDM symbol-by-symbol adaptive transceivers – have a lower packet loss ratio at the same SNR compared to the fixed modulation mode transceiver. Note in Figure 6 that the subband-adaptive transceivers can operate at lower channel SNRs, than the fixed modem mode transceivers, while maintaining the same required video packet loss ratio. Again, the figure labels the subband-adaptive transeivers as μAOFDM, implying that the adaption is not noticable from the upper layers of the system. A macro-adaption could be applied in addition to the microscopic adaption by switching between different target bitrates, as the longer-term channel quality improves and degrades. This issue is the subject of Section 4.6.

Having shown, how the subband-adaptive transceiver achieved a reduced video packet loss, in comparison to fixed modulation mode transceivers under identical channel conditions, we now compare the effective throughput bitrate of the fixed and adaptive OFDM transceivers in Figure 7. The figure shows that when the channel quality is high, the throughput bitrate of the fixed and adaptive transceivers are identical. However, as the channel degrades, the loss of packets results in a lower throughput bitrate. The lower packet loss ratio of the subband-adaptive transceiver results in a higher throughput bitrate than that of the fixed modulation mode transceiver.

The throughput bitrate performance results translate to the decoded video quality performance results evaluated in terms of PSNR in Figure 8. Again, for high channel SNRs, the performance of the fixed and adaptive OFDM transceivers is identical. However, as the channel quality degrades, the video quality of the subband-adaptive transceiver degrades less dramatically, than that of the corresponding fixed modulation mode transceiver.

4.5 Comparing subband-adaptive transceivers having different target bitrates

As mentioned before, the subband-adaptive modems employ different modulation modes for different subcarriers in order to meet the target bitrate requirement at the lowest possible channel SNR. This is achieved by using a more robust modulation mode or eventually by disabling transmissions over subcarriers having a low channel quality. By contrast, the adaptive system can invoke less robust, but higher throughput modulation modes over subcarriers exhibiting a high channel quality. In the examples we have previously considered we chose the AOFDM target bitrate to be identical to that of a fixed modulation mode transceiver. In this section we comparatively study the performance of various μAOFDM systems having different target bitrates.

The previously described μAOFDM transceiver of Table 1 exhibited a FEC-coded bitrate of 7.2Mbps, which was also equivalent to that of a fixed BPSK transceiver and provided an effective video bitrate of 3.4Mbps. If the video target bitrate is lower than 3.4Mbps, then the system can disable transmission in more of the subcarriers, where the channel quality is low. Such a transceiver would have a lower bit error rate, than the previous BPSK-equivalent μAOFDM transceiver, and therefore could be used at lower average channel SNRs, while maintaining the same bit error ratio target. By contrast, as the target bitrate is increased the system has to employ higher-order modulation modes in more subcarriers, at the cost of an increased bit-error ratio. Therefore high target bitrate μAOFDM transceivers can only perform within the required bit error ratio constraints at high channel SNRs, while low target bitrate μAOFDM systems can operate at low channel SNRs without inflicting excessive BERs. Therefore a system, which can adjust its target bitrate, as the channel SNR changes, would operate over a wide range of channel SNRs, providing the maximum possible throughput bitrate, while maintaining the required bit error ratio. Hence below we provide a performance comparison of various μAOFDM transceivers having four different target bitrates, of which two are equivalent to that of the BPSK and QPSK fixed modulation mode transceivers of Table 1. The system parameters for all four different bitrate modes are summarised in Table 2. The modes having effective video bitrates of 3.4 and 7.0Mbps are equivalent to the bitrates of a fixed BPSK and QPSK mode transceiver, respectively.

Figure 9 shows the FER or video packet loss ratio (PLR) performance versus channel SNR for the four different target bitrates of Table 2. The results demonstrate – as expected – that the higher target bitrate modes require higher channel SNRs in order to operate within given PLR constraints. For example, the mode having an effective video bitrate of 9.8Mbps can only operate for channel SNRs in excess of 19dB under the constraint of a maximum PLR of 5%. However, the mode having an effective video bitrate of 3.4Mbps can operate at channel SNRs of 11dB and above whilst maintaining the same 5%

Packet rate	4687.5 Packets/s			
FFT length	512			
OFDM Symbols/Packet	3			
OFDM Symbol Duration	2.6667μs			
OFDM Time Frame	80 Timeslots = 213μs			
Normalised Doppler frequency, f_d'	1.235×10^{-4}			
OFDM symbol normalised Doppler frequency, F_D	7.41×10^{-2}			
FEC Coded Bits/Packet	858	1536	3072	4272
FEC-coded video bitrate	4.0Mbps	7.2Mbps	14.4Mbps	20.0Mbps
No. of Unprotected Bits/Packet	427	766	1534	2134
Unprotected bitrate	2.0Mbps	3.6Mbps	7.2Mbps	10.0Mbps
No. of Error detection CRC (bits)	16	16	16	16
No. of Feedback error flag bits	9	9	9	9
No. of Packet header Bits/Packet	10	11	12	13
Effective video Bits/Packet	392	730	1497	2096
Effective video bitrate	1.8Mbps	3.4Mbps	7.0Mbps	9.8Mbps
Equivalent Modulation mode		BPSK	QPSK	
Minimum channel SNR for 5% PLR (dB)	8.8	11.0	16.1	19.2
Minimum channel SNR for 10% PLR (dB)	7.1	9.2	14.1	17.3

Table 2: System parameters for the four different target bitrates of the various subband-adaptive OFDM transceivers (μAOFDM).

PLR constraint, albeit at about half the throughput bitrate, and hence at a lower video quality.

The tradeoffs between video quality and channel SNR for the various target bitrates can be judged from Figure 10, suggesting – as expected – that the higher target bitrates result in a higher video quality, provided that channel conditions are sufficiently favorable. However, as the channel quality degrades, the video packet loss ratio increases, thereby reducing the throughput bitrate, and hence the associated video quality. The lower target bitrate transceivers operate at an inherently lower video quality, but they are more robust to the prevailing channel conditions and hence can operate at lower channel SNRs, while guaranteeing a video quality, which is essentially unaffected by channel errors. It was found that the perceived video quality became impaired for packet loss ratios in excess of about 5%.

The tradeoffs between video-quality, packet loss ratio and the target bitrate are further augmented with reference to Figures 11(a), (b) and (c). The figure shows the video quality measured in PSNR versus video frame index at a channel SNR of 16dB and also for an error free situation. At the bottom of each graph the packet loss ratio per video frame is shown. The three figures indicate the tradeoffs to be made in choosing the target bitrate for the specific channel conditions experienced – in this specific example for a channel SNR of 16dB. Note that under error free conditions the video quality improved upon increasing the bitrate.

Specicially, video PSNRs of about 40, 41.5 and 43dB were observed for the effective video bitrates of 1.8, 3.4 and 7.0Mbps. Figure 11(a) shows that for the target bitrate of 1.8Mbps, the system has a high grade of freedom in choosing, which subcarriers to invoke and therefore it is capable of reducing the number of packets that are lost. The packet loss ratio remains low and the video quality remains similar to that of the error free situation. The two instances, where the PSNR is significantly different from the error free performance correspond to video frames, in which video packets were lost. However, the system recovers in both instances in the following video frame.

As the target bitrate of the subband-adaptive OFDM transceiver is increased to 3.4Mbps (see Figure 11(b)), the subband modulation mode selection process has to be more "aggressive", resulting in increased video packet loss. Observe in the figure that the transceiver having an effective video bitrate of 3.4Mbps, exhibits increased packet loss, and in one frame as much as 5% of the packets transmitted for that video frame were lost, although the average PLR was only 0.4%. Due to the increased packet loss the video PSNR curve diverges from the error-free performance curve more often. However, in almost all cases the effects of the packet losses are masked in the next video frame, indicated by the re-merging PSNR curves in the figure, maintaining a close to error-free PSNR. The subjective effect of this level of packet loss is almost inperceivable.

When the target bitrate is further increased to 7.0Mbps (see Figure 11(c)), the average PLR is about 5%

under the same channel conditions, and the effects of this packet loss ratio are becoming objectionable in perceived video quality terms. At this target bitrate, there are several video frames, where at least 10% of the video packets have been lost. The video quality measured in PSNR terms rarely reaches its error-free level, due to the fact that every video frame contains at least one lost packet. The perceived video quality remains virtually unimpaired, until the head movement in the "Suzie" video sequence around frames 40–50, where the effect of lost packets becomes obvious, and the PSNR drops to about 30dB.

4.6 Modifying the target bitrate based on channel conditions

By using a high target bitrate, when the channel quality is high, while a reduced target bitrate, when the channel quality is poor, such an adaptive system is capable of maximising the average throughput bitrate over a wide range of channel SNRs, while maintaining a given quality constraint. This quality constraint for our video system could be a maximum packet loss ratio.

However there is a substantial processing delay associated with evaluating the packet loss information and therefore modem mode switching based on this metric would be less efficient due to this latency. Therefore we decided to invoke an estimate of the bit error ratio (BER) for mode switching. The channel quality estimator can estimate the expected bit error ratio based on each specific modulation mode chosen for each subband. We decided to use a quadruple-mode switched subband-adaptive modem, using the four target bitrates of Table 2. The channel estimator can then estimate the expected bit error ratio of the four possible modem modes. The modem mode for the next OFDM symbol is then choosen based upon the estimate of BER for each of the four modes. Our switching scheme opted for the modem mode, whose estimated BER was below the required threshold. This threshold could be varied in order to tune the behaviour of the switched subband-adaptive modem for a high or a low throughput. The advantage of a higher throughput was a higher error-free video quality at the expense of increased video packet losses, which could reduce the perceived video quality.

Figure 12 demonstrates, how the switching algorithm operates for a 1% estimated BER threshold. Specifically, the figure portrays the estimate of the bit error ratio for the four possible modem modes versus time. The large square and the dotted line indicates the mode chosen for each time interval by the mode switching algorithm. The algorithm attempts to use the highest bitrate mode, whose BER estimate is less than the target threshold namely, 1% in this case. However, if all the four modes' estimate of the BER is above the 1% threshold, then the lowest bitrate mode is chosen, since this will be the most robust to channel errors. An example of this is shown around frames 1035–1040. At the bottom of the graph a bar chart specifies the bitrate of the switched subband adaptive modem versus time, in order to emphasise when the switching occurs.

An example of the algorithm when switching amongst the target bitrates of 1.8, 3.4, 7 and 10Mbps is shown in Figures 13(a) and (b). Figure 13(a) portrays the contour plot of the channel SNR for each subcarrier versus time. Figure 13(b) displays the modulation mode chosen for each 32-subcarrier subband versus time for the time-varient target bitrate (TVTBR) subband adaptive modem. It can be seen at frames 1051–1055 that all the subbands employ QPSK modulation, therefore the TVTBR-AOFDM modem has an instantaneous target bitrate of 7Mbps. As the channel used by the 3.4 Mbps QPSK mode degrades around frame 1060, the modem has switched to the more robust 1.8Mbps BPSK mode. When the channel quality is high around frames 1074-1081, the highest bitrate 10Mbps 16QAM mode is used. This demonstrates that the TVTBR-AOFDM modem, can reduce the number of lost video packets, by using reduced bitrate but more robust modulation modes, when the channel quality is poor. However, this is at the expense of a slightly reduced average throughput bitrate. Usually a higher througput bitrate results in a higher video quality, however a high bitrate associated with a high packet loss ratio, is usually less attractive in terms of perceived video quality than a lower bitrate, lower packet loss ratio mode.

Having highlighted how the time-domain mode switching algorithm operates, we will now characterise its performance for a range of different BER switching thresholds. A low BER switching threshold implies that the switching algorithm is cautious about switching to the higher bitrate modes, and therefore the system performance is characterised by a low video packet loss ratio, and a low throughput bitrate. A high BER switching threshold results in the switching algorithm attempting to use the highest bitrate modes in all but the worst channel conditions. This results in a higher video packet loss ratio. However, if the packet loss ratio is not excessively high, a higher video throughput is achieved.

Figure 14 portrays the video packet loss ratio or FER performance of the TVTBR-AOFDM modem for a variety of BER thresholds, compared to the minimum and maximum rate un-switched modes. It can be seen that for a conservative BER switching threshold of 0.1% the time-varient target bitrate subband adaptive (TVTBR-AOFDM) modem has a similar packet loss ratio performance to that of the 1.8Mbps non-switched or constant target bitrate (CTBR) subband adaptive modem. However, as we will show, the throughput of the switched modem is always better or equal to that of the un-switched modem, and becomes far superior, as the channel quality improves. Observe in the figure that the "agressive" switching threshold of 10% has a similar packet loss ratio performance to that of the 9.8Mbps CTBR-AOFDM modem. We found that in order to maintain a packet loss ratio of below 5%, the BER switching thresholds of 2 and 3% offered the best overall performance, since the packet loss ratio was fairly low, while the throughput bitrate was higher, than that of an un-switched CTBR-AOFDM modem.

A high BER switching threshold results in the switched subband adaptive modem transmitting at a high average bitrate. However, we have shown in Figure 14 how the packet loss ratio increases, as the BER switching threshold is increased. Therefore the overall useful or effective throughput bitrate, ie. the bitrate excluding lost packets, can be reduced in conjunction with high BER switching thresholds.

Figure 15 demonstrates how the transmitted bitrate of the switched TVTBR-AOFDM modem increases with higher BER switching thresholds. However, when this is compared to the effective throughput bitrate, where the effects of packet loss are taken into account, the tradeoff between the BER switching threshold and the effective bitrate is less apparent.

Figure 16 portrays the corresponding effective throughput bitrate versus channel SNR for a range of BER switching thresholds. The figure demonstrates that for a BER switching threshold of 10% the effective throughput bitrate performance was reduced incomparison to some of the lower BER switching threshold scenarios. Therefore the BER=10% switching threshold is clearly too aggressive, resulting in a high packet loss ratio, and a reduced effective throughput bitrate. For the switching thresholds considered, the BER=5% threshold achieved the highest effective throughput bitrate. However, even though the BER=5% switching threshold produces the highest effective throughput bitrate, this is at the expense of a relatively high video packet loss ratio, which – as we will show – has a detrimental effect on the perceived video quality.

We will now demonstrate the effects associated with different BER switching thresholds on the video quality represented by the peak-signal-to-noise ratio (PSNR).

Figures 17(a)-17(c) portray the PSNR and packet loss performance versus time for a range of BER switching thresholds.

Figure 17(a) indicates that for a BER switching threshold of 1% the PSNR performance is very similar to the corresponding error-free video quality. However, the PSNR performance diverges from the error-free curve, when video packets are lost, although the highest PSNR degradation is limited to 2dB. Furthermore, the PSNR curve typically reverts to the error-free PSNR performance curve in the next frame. In this example about 80% of the video frames have no video packet loss.

When the switching threshold is increased to 2%, as shown in Figure 17(b), the video packet loss ratio has increased, such that now only 41% of video frames have no packet loss. The result of the increased packet loss is a PSNR curve, which diverges from the error-free PSNR performance curve more regularly, with PSNR degradations of upto 7dB. It is worth noting that when there are video frames with no packet losses, the PSNR typically recovers, achieving a similar PSNR performance to the error-free case. When the BER switching threshold was further increased to 3%, which is not shown in the figure, the maximum PSNR degradation increased to 10.5dB, and the number of video frames without packet losses was reduced to 6%.

Figure 17(c) portrays the PSNR and packet loss performance for a BER switching threshold of 5%. The

PSNR degradation in this case ranges from 1.8 to 13dB and all video frames contain at least one lost video packet. Even though the BER=5% switching threshold provides the highest effective throughput bitrate, the associated video quality is poor. The PSNR degradation in most video frames is about 10dB. Clearly, the highest effective throughput bitrate does not guarantee the best video quality. We will now demonstrate that the switching threshold of BER=1% provides the best video quality, when using the average PSNR as a performance metric.

Figure 18(a) compares the average PSNR versus channel SNR performance for a range of switched (TVTBR) and un-switched (CTBR) AOFDM modems. The figure compares the four un-switched, ie. CTBR subband adaptive modems with switching, ie. TVTBR subband adaptive modems, which switch between the four fixed-rate modes, depending on the BER switching threshold. The figure indicates that the switched TVTBR subband adaptive modem having a switching threshold of BER=10% results in similar PSNR performance to the un-switched CTBR 9.8Mbps subband adaptive modem. When the switching threshold is reduced to BER=3%, the switched TVTBR AOFDM modem outperforms all of the un-switched CTBR AOFDM modems. A switching threshold of BER=5% achieves a PSNR performance, which is better than the un-switched 9.8Mbps CTBR AOFDM modem, but worse than the un-switched 7.0Mbps modem, at low and medium channel SNRs.

A comparsion of the switched TVTBR AOFDM modem employing all six switching thresholds that we have used previously is shown in Figure 18(b). This figure suggests that switching thresholds of BER=0.1, 1 and 2% perform better than the BER=3% threshold, which outperformed all of the un-switched CTBR subband adaptive modems. The best average PSNR performance was acheieved by a switching threshold of BER=1%. The more conservative BER=0.1% switching threshold results in a lower PSNR performance, since its throughput bitrate was significantly reduced. Therefore the best tradeoff in terms of PSNR, throughput bitrate and video packet loss ratio was achieved with a switching threshold of about BER=1%.

4.7 Summary of Embodiments

We have outlined an adaptive modulation technique, which can be applied to OFDM systems. In practical terms the subcarrier modem mode cannot be independently chosen for each subcarrier, since the associated modem mode side-information would be prohibitively high. Hence we divided the subcarriers into subband and controlled the modulation modes on a subband-by-subband basis, which resulted in an acceptable side information requirement.

In Section 4.4 we compared the performance of subband adaptive OFDM modems to conventional OFDM modems, operating at the same bitrate. The subband adaptive modem could invoke BPSK, QPSK, or 16QAM modulation for each subband, or disable transmission for a subband, if the channel conditions were poor. The subband adaptive modems could provide a lower BER, than the corresponding convention BPSK or QPSK OFDM modems at the same channel SNR. This was acheived by transmitting more bits in the higher-quality subbands, and less bits in the lower-quality subbands, thereby reducing the chances of corrupted bits. The lower BER of the subband adaptive OFDM modems provided a higher effective video bitrate for the video codec, which in turn provided a higher video quality. Additionally the subband adaptive modem could operate at lower channel SNRs, while maintaining the required video quality.

In Section 4.5 we compared the performance of subband adaptive OFDM modems, operating at different target bitrates. This showed that higher target bitrates required a higher channel quality. This was further exploited in Section 4.6, where we added another level of adaption, by switching between different target bitrates, based on the prevailing channel conditions. This enabled a time-variant target bitrate subband adative OFDM (TVTBR-AOFDM) modem to provide a higher bitrate, when the overall channel quality was high, and a lower bitrate when the overall channel quality was poor, in order to maintain the required video quality.

The proposed constant target bitrate subband adaptive OFDM (CTBR-AOFDM) modems can provide a lower BER, than a corresponding conventional OFDM modem. The slightly more complex switched TVTBR-AOFDM modems can provide a balanced video quality performance, across a wider range of channel SNRs.

4.8 Conclusions

The proposed burst-by-burst adaptive multimedia OFDM transceiver concept exhibits substantial advantages in comparison to conventional fixed-mode OFDM transceivers, which was substantiated in the context of a specific embodiment of the advocated system concept, namely with the aid of a burst-by-burst adaptive video transceiver.

Specifically, the main advantage of the proposed burst-by-burst adaptive OFDM multimedia transceiver technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video, speech or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, etc. Furthermore, when the mobile is roaming in a hostile outdoor propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

The proposed system concept has the following important features:

1. A reliable near-instantaneous channel quality metric is employed, in order to appropriately configure the AOFDM modem for maintaining the required target BER and the associated source signal representation quality.

2. The perceived channel quality determines the number of bits that can be transmitted in a given OFDM transmission burst, which in turn predetermines the number of bits to be generated by the associated multimedia source codec, such as for example the associated video, audio or speech codec. Hence the multimedia source codec has to be capable of adjusting the number of bits generated under the instruction of the burst-by-burst adaptive OFDM transceiver.

3. The OFDM transmitter mode requested by the receiver, in order to achieve the target performance has to be signalled by the receiver to the remote transmitter. Another scenario is, where the uplink and downlink channel quality is sufficiently similar for allowing the receiver to judge, what transmission mode the associated transmitter should use, in order for its transmitted signal to mantain the required transmission integrity. Lastly, the mode of operation used by the transmitter can also be detected using blind detection techniques, for example in conjunction with the associated channel decoder.

4. In practical terms the AOFDM subcarrier modem mode cannot be independently chosen for each subcarrier, since the associated modem mode side-information would be prohibitively high. Hence we proposed to divide the AOFDM subcarriers into subbands and to control the modulation modes on a subband-by-subband basis, which resulted in an acceptable side information requirement.

5. The subband adaptive modems may provide a lower BER, than the corresponding conventional BPSK or QPSK OFDM modems at the same channel SNR. This was achieved by transmitting more bits in the higher-quality subbands, and less bits in the lower-quality subbands, thereby reducing the chances of corrupted bits. The lower BER of the subband adaptive OFDM modems provided a higher effective video bitrate for the video codec in the studied embodiment of the proposed system, which in turn provided a higher video quality. Additionally the subband adaptive modem could operate at lower channel SNRs, while maintaining the required video quality.

6. Higher AOFDM target bitrates required a higher channel quality. This was further exploited in Section 4.6, where we added another level of adaption, by switching between different target

bitrates, based on the prevailing channel conditions. This enabled a time-variant target bitrate subband adative OFDM (TVTBR-AOFDM) modem to provide a higher bitrate, when the overall channel quality was high, and a lower bitrate when the overall channel quality was poor, in order to maintain the required video quality.

7. The proposed constant target bitrate subband adaptive OFDM (CTBR-AOFDM) modems can provide a lower BER, than a corresponding conventional OFDM modem. The slightly more complex switched TVTBR-AOFDM modems can provide a balanced video quality performance, across a wider range of channel SNRs.

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CLAIMS

1. A transmitter for transmission of a multimedia source signal over a transmission medium to a remote receiver, the transmitter comprising:

an AOFDM modem having an output for transmitting a multimedia source signal;

a source codec arranged to supply the multimedia source signal to the modem; and

an input for receiving a metric of channel quality indicative of current transmission integrity;

wherein the modem and/or the source codec are reconfigurable responsive to the channel quality in order to maintain a required multimedia source signal integrity at a remote receiver receiving the multimedia source signal transmitted by the modem.

2. A transmitter according to claim 1, wherein the required multimedia source signal integrity is defined in terms of a target bit error rate (BER) and/or a target AOFDM symbol error rate (SER) at the remote receiver.

3. A transmitter according to claim 2, further comprising a channel quality estimator for determining BER and/or SER estimates for each of a plurality of transmission modes of the modem, the modem being operable according to a switching scheme whereby the transmission mode is chosen based upon the BER or SER estimates for the transmission modes.

4. A transmitter according to claim 3, wherein the transmission mode is chosen to be the transmission mode with the highest bit rate that has an estimate of the target BER or target AOFDM SER below a threshold.

5. A transmitter according to claim 4, wherein the threshold target BER or SER is variable according to prevailing channel conditions.

6. A transmitter according to any one of claims 3 to 5, wherein the AOFDM modem is operable to transmit using a plurality of subcarrier subbands, the transmission mode being independently selectable for the different subbands, whereby higher transmission rates are achieved in higher-quality subbands and lower transmission rates in lower-quality subbands.

7. A transmitter according to any one of claims 2 to 6, wherein the target BER is set to limit the AOFDM SER to a maximum.

8. A transmitter according to any one of the preceding claims, wherein the required multimedia source signal integrity is defined in terms of an AOFDM SER at the remote receiver.

9. A transmitter according to any one of the preceding claims, wherein the modem is reconfigurable in use by varying the number of bits per AOFDM symbol responsive to the channel quality.

10. A transmitter according to any one of the preceding claims, wherein the channel quality metric is based on the current BER or AOFDM SER detected at the remote receiver and transmitted back to the transmitter.

11. A transmitter according to any one of claims 1 to 10, wherein the channel quality metric is based on the current transmission BER or AOFDM SER detected at a receiver local to the transmitter sharing the transmitter's transmission medium.

12. A transmission system for transmission of multimedia source signals over a transmission medium, the system comprising:

a first transceiver including a local receiver and a local transmitter according to any one of the preceding claims; and

a second transceiver including a remote receiver and a remote transmitter according to any one of the preceding claims.

13. A method of transmitting a multimedia source signal, the method comprising:

providing a transmitter comprising a source encoder and AOFDM modulator;

generating a multimedia source signal in the source encoder;

supplying the multimedia source signal to the AOFDM modulator;

transmitting the multimedia source signal from the AOFDM modulator over a transmission medium to a remote receiver;

obtaining a channel quality metric indicative of channel quality experienced by the receiver; and

controlling the source encoder and/or the AOFDM modulator responsive to the channel quality metric so that the integrity of the signal received by the receiver meets a desired integrity target.

14. A method according to claim 13, wherein the desired integrity target is defined in terms of a bit error rate (BER) and an AOFDM symbol error rate (SER).

15. A method according to claim 13 or 14, wherein the modem is switched between a plurality of transmission modes according to an estimate of the expected BER or AOFDM SER for the individual transmission modes obtained on the basis of the estimated channel quality, thereby selecting the transmission mode having the highest transmission rate that complies with the desired integrity target.

16. A method according to claim 15, wherein the AOFDM modulator transmits using a plurality of subcarrier subbands, the transmission modes being independently selected for the individual subbands.

17. A method according to any one of claims 13 to 16, wherein the source encoder is reconfigured during transmission according to the number of bits per AOFDM symbol to be generated responsive to the channel quality metric.

18. A method according to any one of claims 13 to 17, wherein the channel quality metric is estimated from the signal received at the receiver and transmitted

back to the transmitter via a feedback path to set the modulation mode of the transmitter to meet the desired integrity target at the receiver.

19. A method according to any one of claims 13 to 17, wherein the channel quality metric is estimated from signals transmitted from a remote transmitter over the transmission medium to a local receiver.

20. A method according to any one of claims 13 to 19, wherein the channel quality predetermines the source-representation quality of the multimedia source signal received by the receiver under error-free channel conditions.

Fig. 1(a)

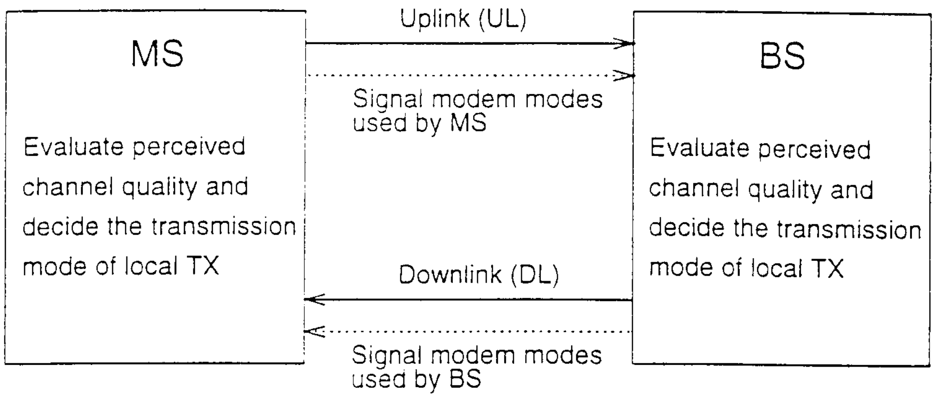


Fig. 1(b)

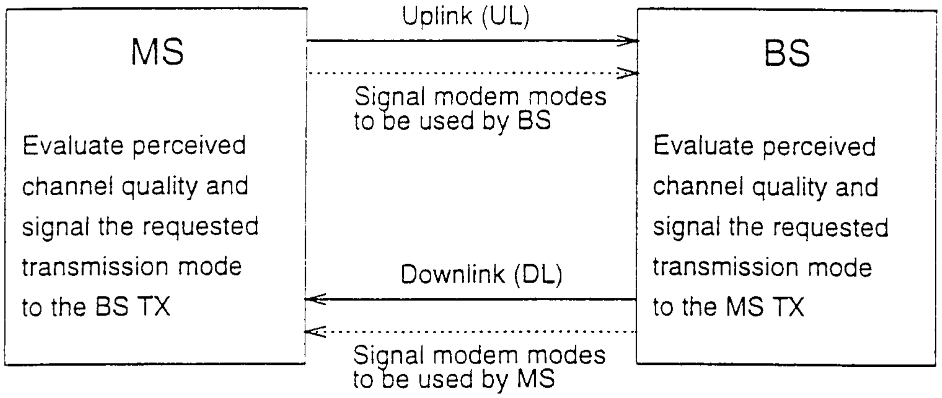


Fig. 1(c)

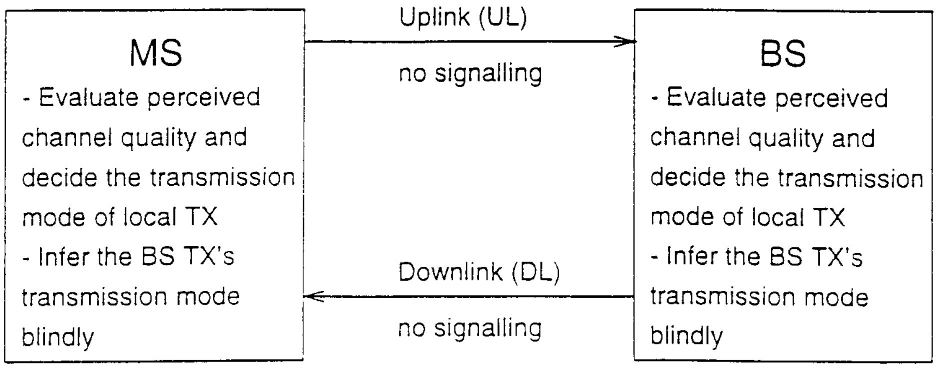


Fig. 2

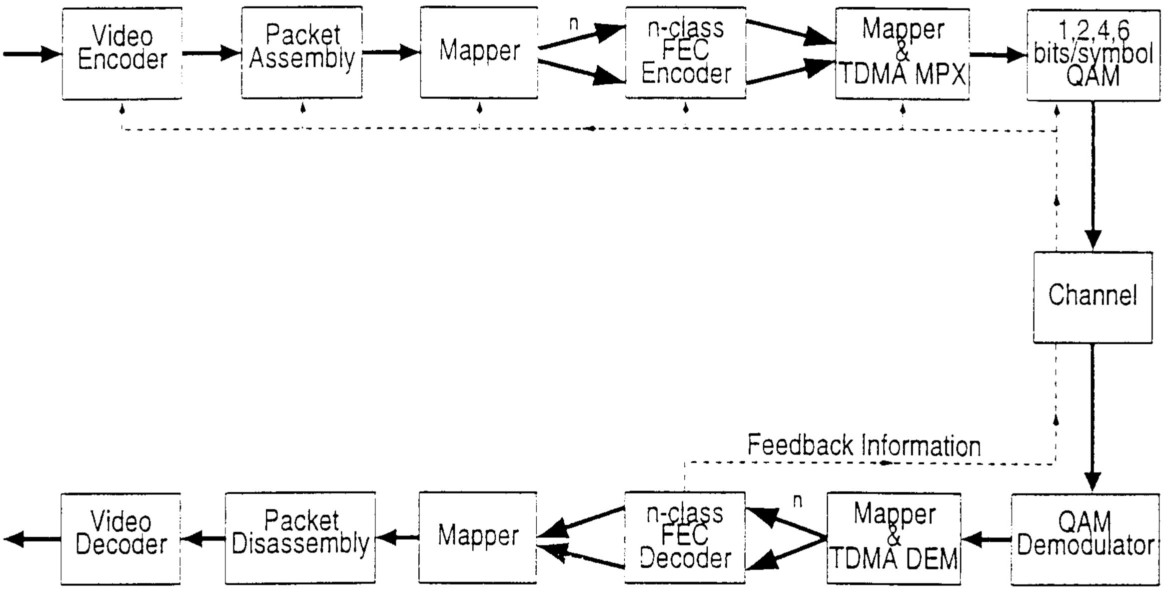

Video Encoder
Packet Assembly
Mapper
n
n-class FEC Encoder
Mapper & TDMA MPX
1,2,4,6 bits/symbol QAM
Channel
Feedback Information
Video Decoder
Packet Disassembly
Mapper
n-class FEC Decoder
n
Mapper & TDMA DEM
QAM Demodulator

Fig. 3(a)

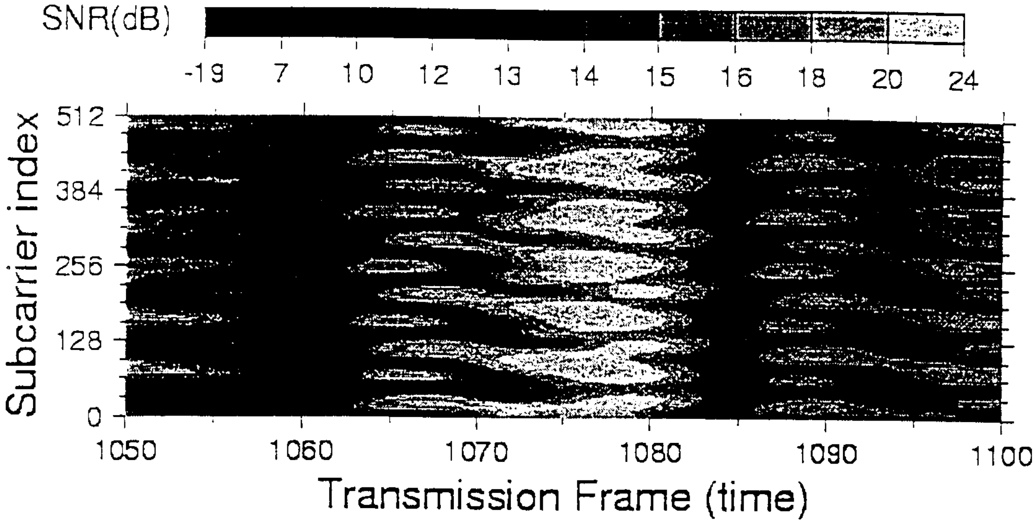


Fig. 3(b)

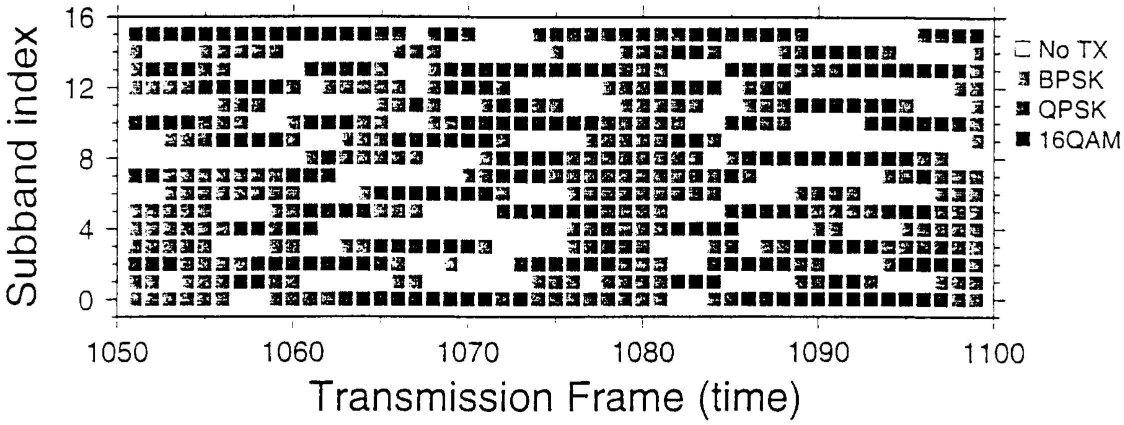


Fig. 3(c)

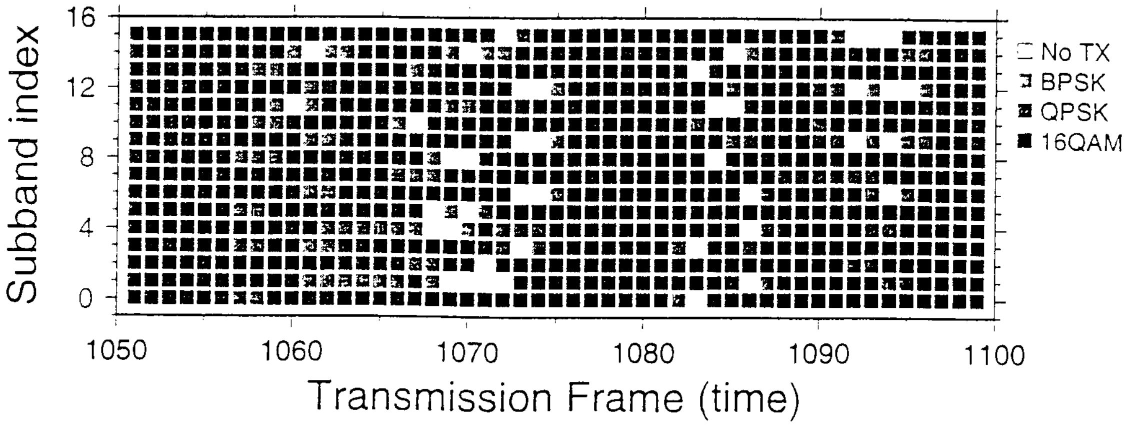


Fig. 4

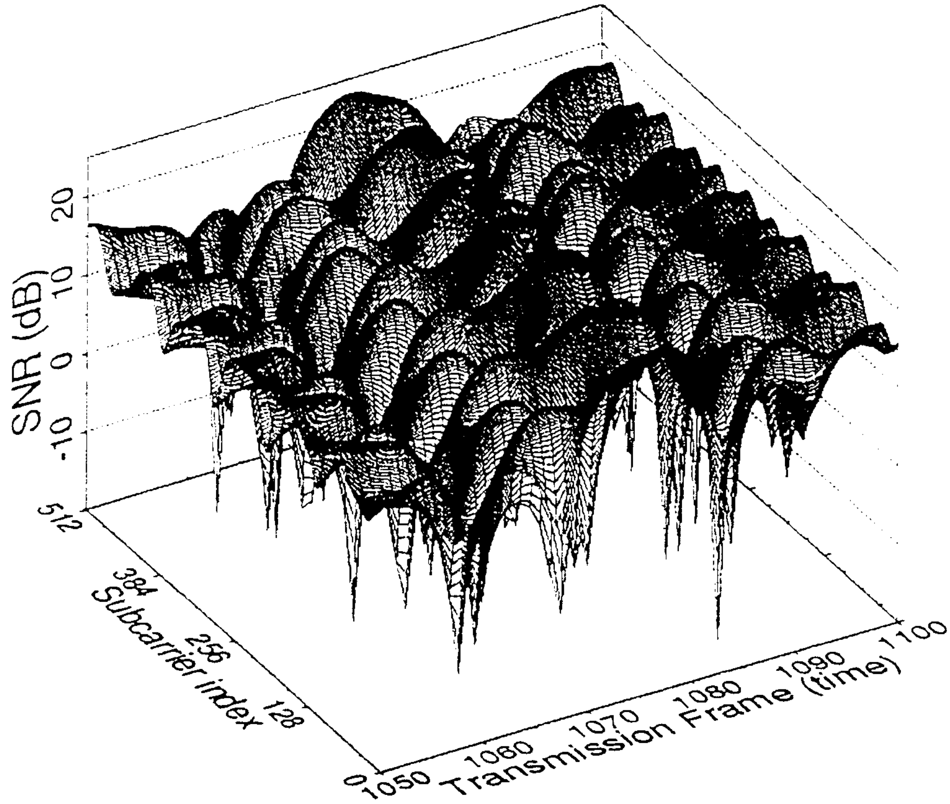

SNR (dB)
20
10
0
-10
512
384
256
128
0
Subcarrier index
1050
1060
1070
1080
1090
1100
Transmission Frame (time)

Fig. 5

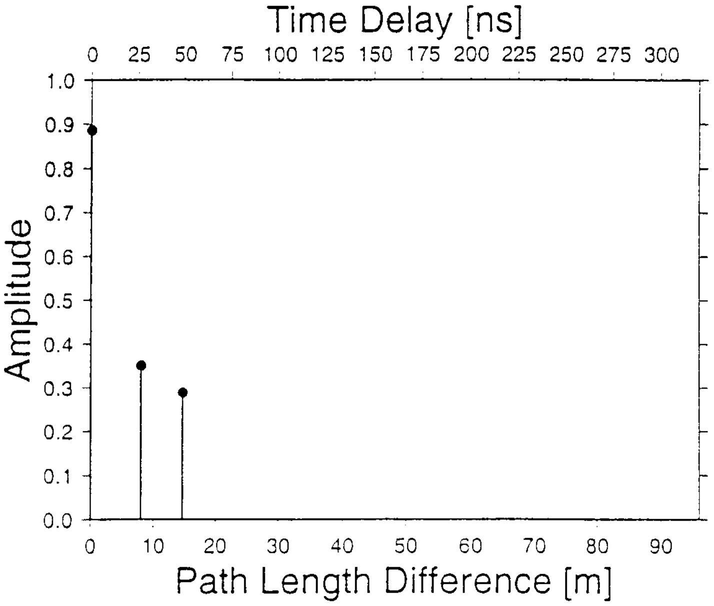

Time Delay [ns]
0
25
50
75
100
125
150
175
200
225
250
275
300
Amplitude
1.0
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0.0
0
10
20
30
40
50
60
70
80
90
Path Length Difference [m]

Fig. 6

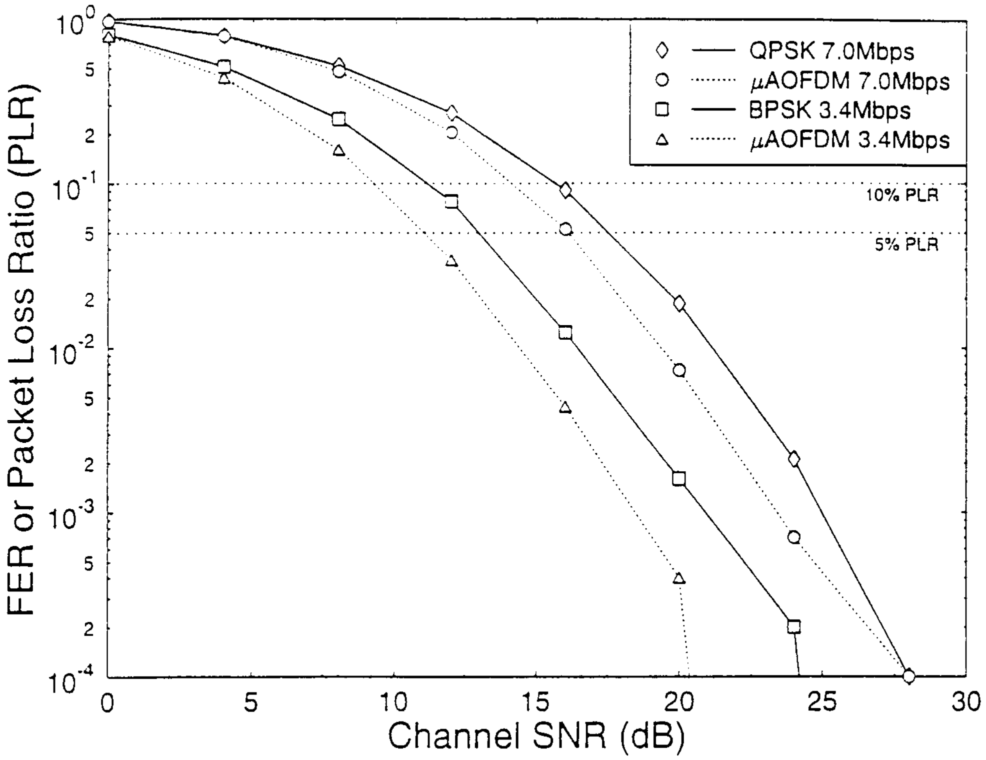

10^0
5
2
10^{-1}
5
2
10^{-2}
5
2
10^{-3}
5
2
10^{-4}
FER or Packet Loss Ratio (PLR)
0
5
10
15
20
25
30
Channel SNR (dB)
◇ —— QPSK 7.0Mbps
○ ········ μAOFDM 7.0Mbps
□ —— BPSK 3.4Mbps
△ ········ μAOFDM 3.4Mbps
10% PLR
5% PLR

Fig. 7

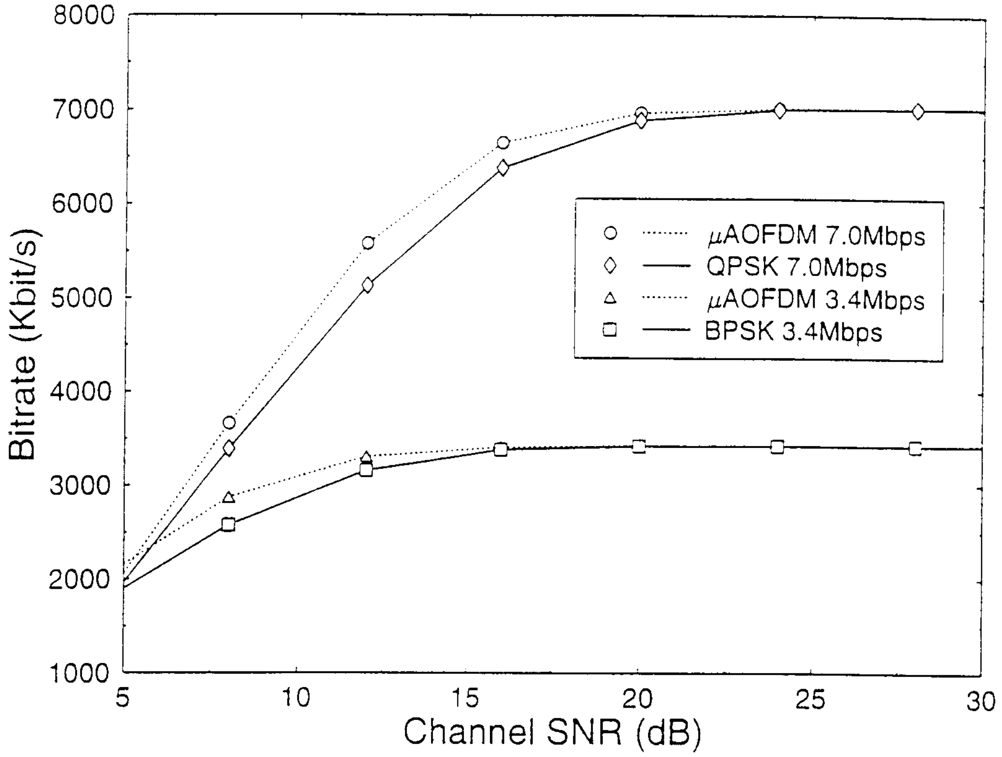

8000
7000
6000
5000
4000
3000
2000
1000
Bitrate (Kbit/s)
5
10
15
20
25
30
Channel SNR (dB)
○ μAOFDM 7.0Mbps
◇ QPSK 7.0Mbps
△ μAOFDM 3.4Mbps
□ BPSK 3.4Mbps

Fig. 8

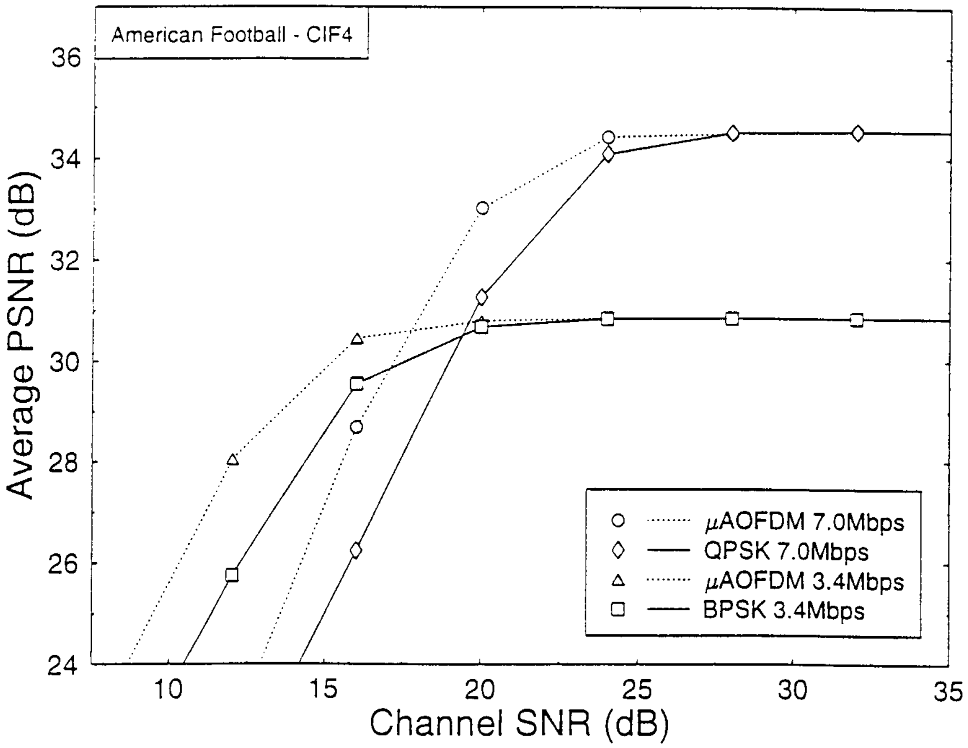

American Football - CIF4
Average PSNR (dB)
24
26
28
30
32
34
36
Channel SNR (dB)
10
15
20
25
30
35
○ μAOFDM 7.0Mbps
◇ QPSK 7.0Mbps
△ μAOFDM 3.4Mbps
□ BPSK 3.4Mbps

Fig. 9

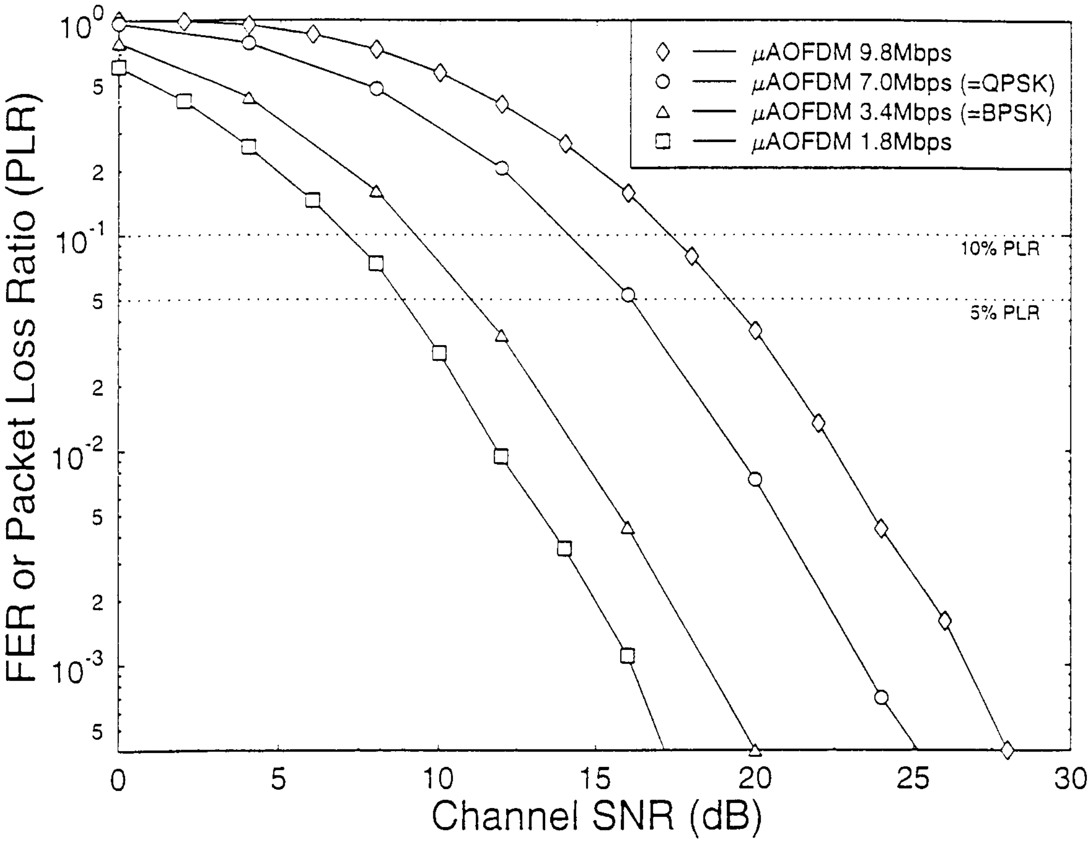

◊ —— μAOFDM 9.8Mbps
○ —— μAOFDM 7.0Mbps (=QPSK)
△ —— μAOFDM 3.4Mbps (=BPSK)
□ —— μAOFDM 1.8Mbps
10% PLR
5% PLR
FER or Packet Loss Ratio (PLR)
10^0
5
2
10^{-1}
5
2
10^{-2}
5
2
10^{-3}
5
0
5
10
15
20
25
30
Channel SNR (dB)

Fig. 10

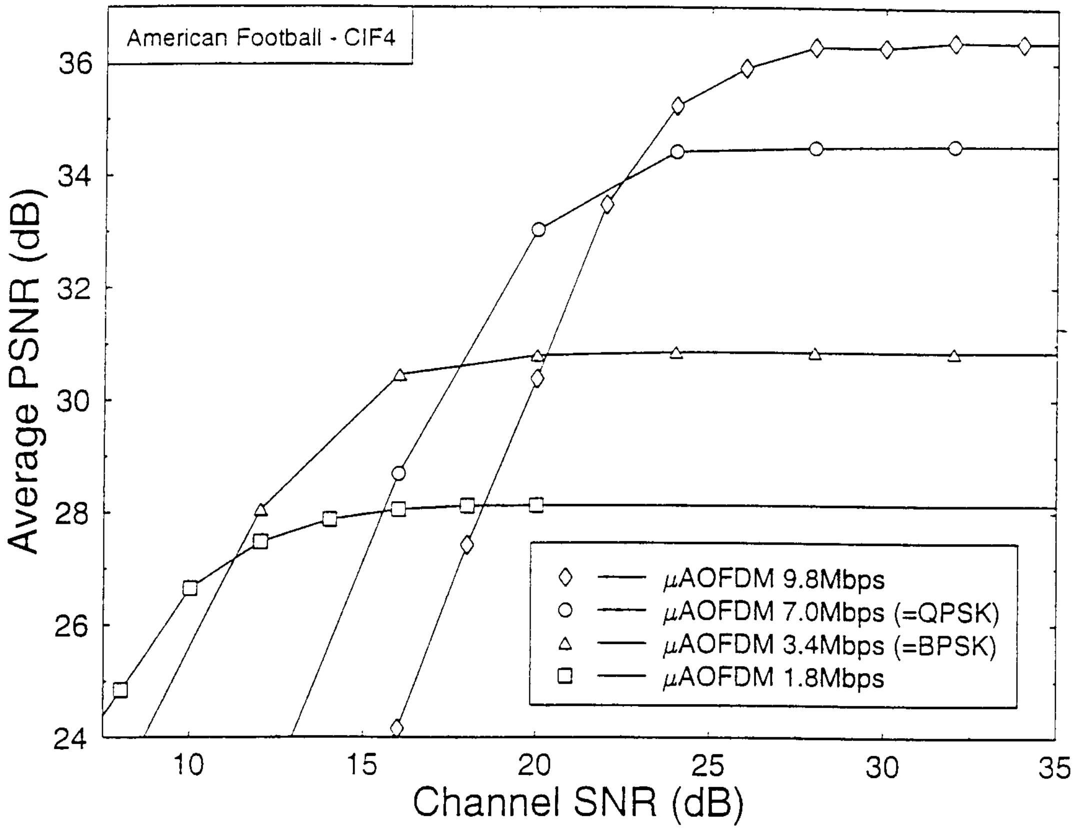

American Football - CIF4
Average PSNR (dB)
36
34
32
30
28
26
24
10
15
20
25
30
35
Channel SNR (dB)
◇ — μAOFDM 9.8Mbps
○ — μAOFDM 7.0Mbps (=QPSK)
△ — μAOFDM 3.4Mbps (=BPSK)
□ — μAOFDM 1.8Mbps

Fig. 11(a)

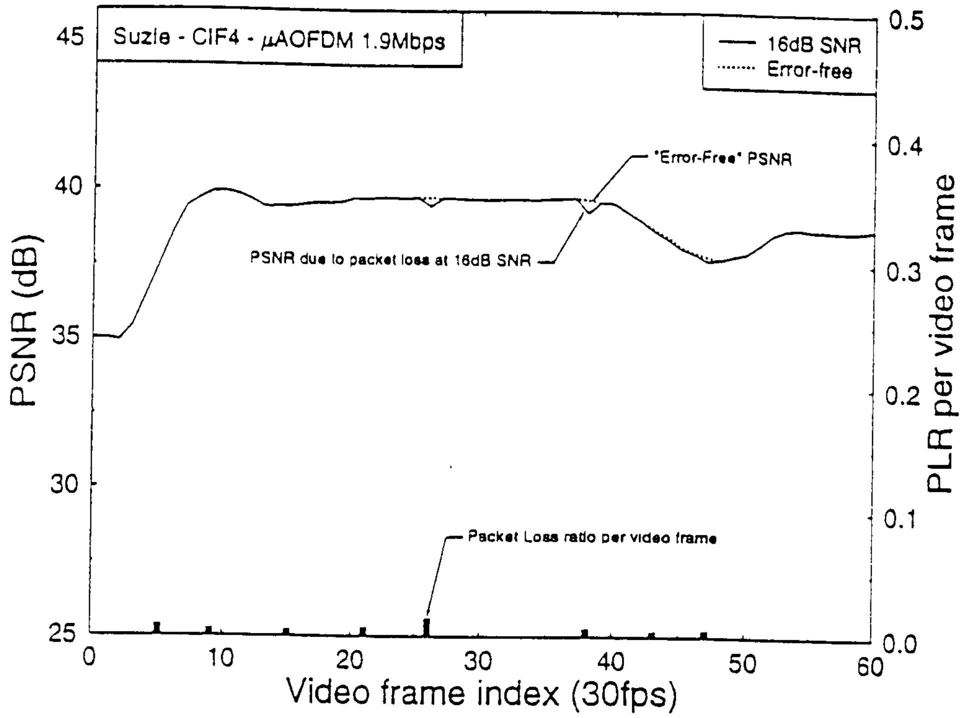


Fig. 11(b)

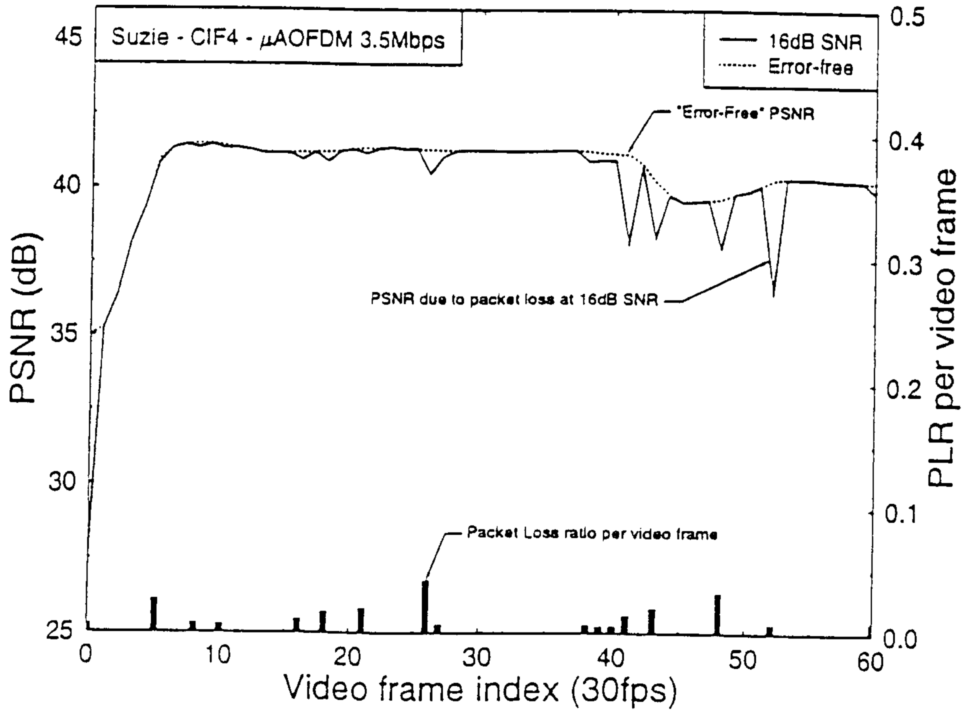


Fig. 11(c)

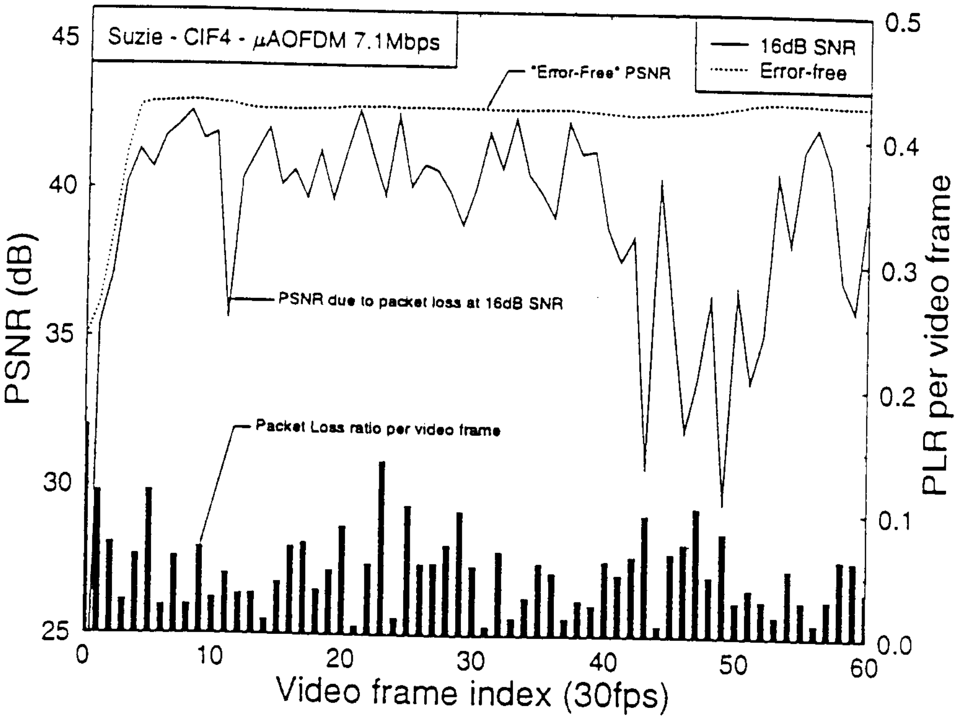

Suzie - CIF4 - μAOFDM 7.1Mbps
16dB SNR
Error-free
"Error-Free" PSNR
PSNR due to packet loss at 16dB SNR
Packet Loss ratio per video frame
PSNR (dB)
45
40
35
30
25
PLR per video frame
0.5
0.4
0.3
0.2
0.1
0.0
0
10
20
30
40
50
60
Video frame index (30fps)

Fig. 12

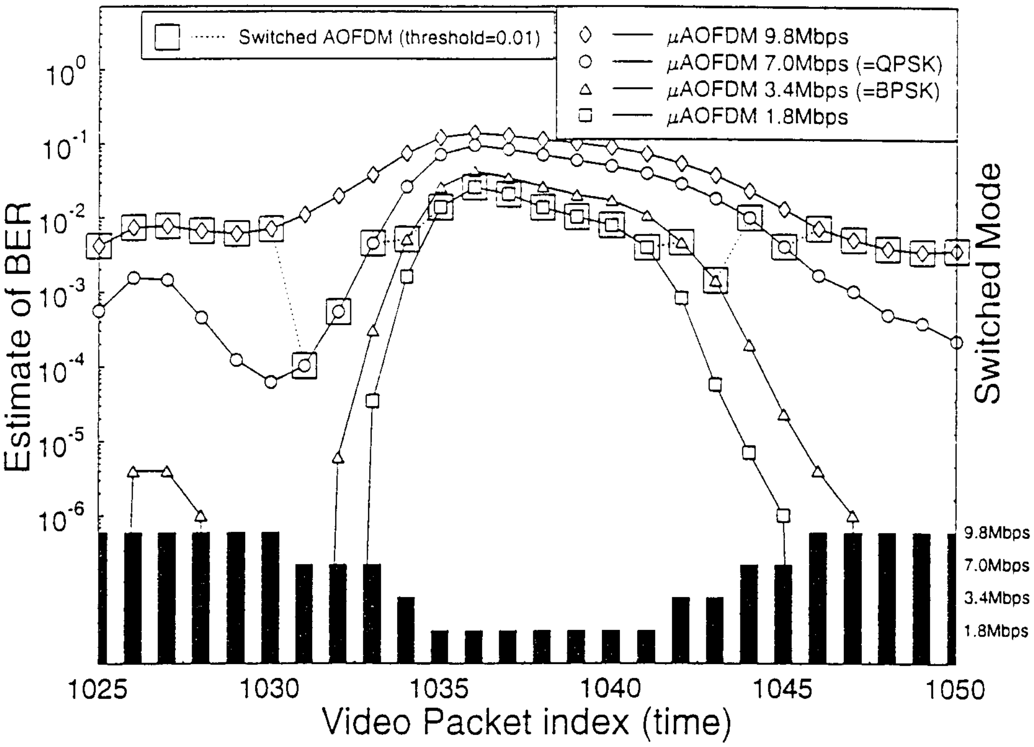

Switched AOFDM (threshold=0.01)
μAOFDM 9.8Mbps
μAOFDM 7.0Mbps (=QPSK)
μAOFDM 3.4Mbps (=BPSK)
μAOFDM 1.8Mbps
Estimate of BER
10^0
10^{-1}
10^{-2}
10^{-3}
10^{-4}
10^{-5}
10^{-6}
Switched Mode
9.8Mbps
7.0Mbps
3.4Mbps
1.8Mbps
1025
1030
1035
1040
1045
1050
Video Packet index (time)

Fig. 13(a)

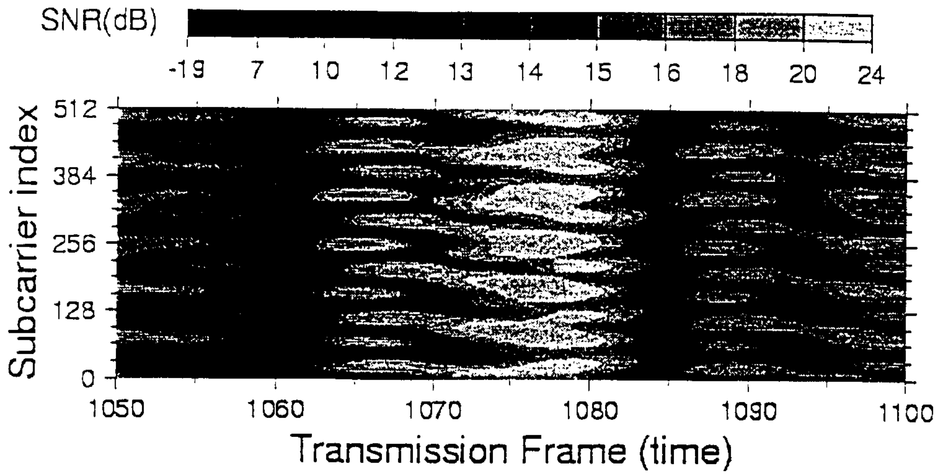


Fig. 13(b)

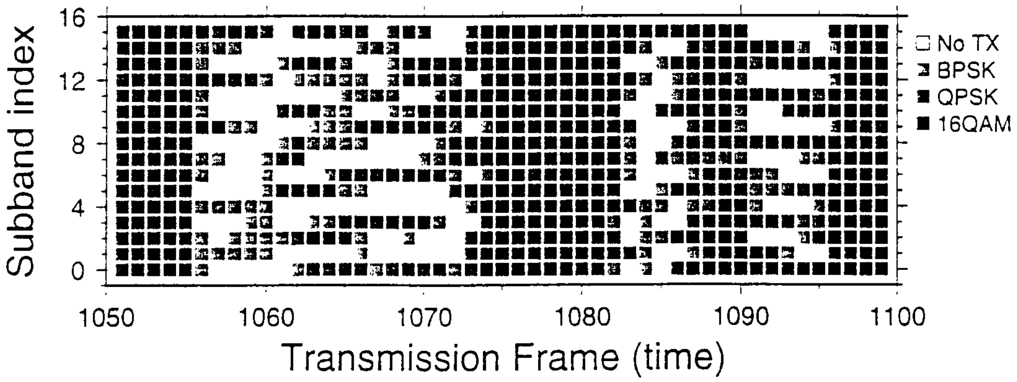


Fig. 14

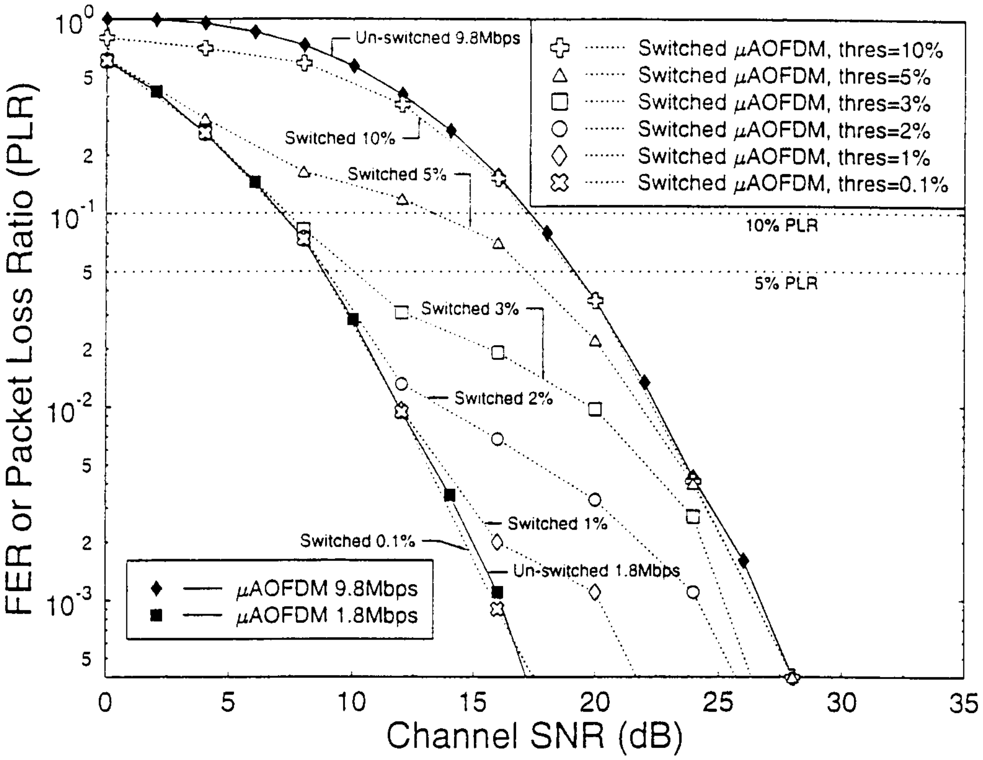

FER or Packet Loss Ratio (PLR)
Channel SNR (dB)
10^0
10^{-1}
10^{-2}
10^{-3}
5
2
0
5
10
15
20
25
30
35
Un-switched 9.8Mbps
Switched 10%
Switched 5%
Switched 3%
Switched 2%
Switched 1%
Switched 0.1%
Un-switched 1.8Mbps
10% PLR
5% PLR
Switched μAOFDM, thres=10%
Switched μAOFDM, thres=5%
Switched μAOFDM, thres=3%
Switched μAOFDM, thres=2%
Switched μAOFDM, thres=1%
Switched μAOFDM, thres=0.1%
μAOFDM 9.8Mbps
μAOFDM 1.8Mbps

Fig. 15

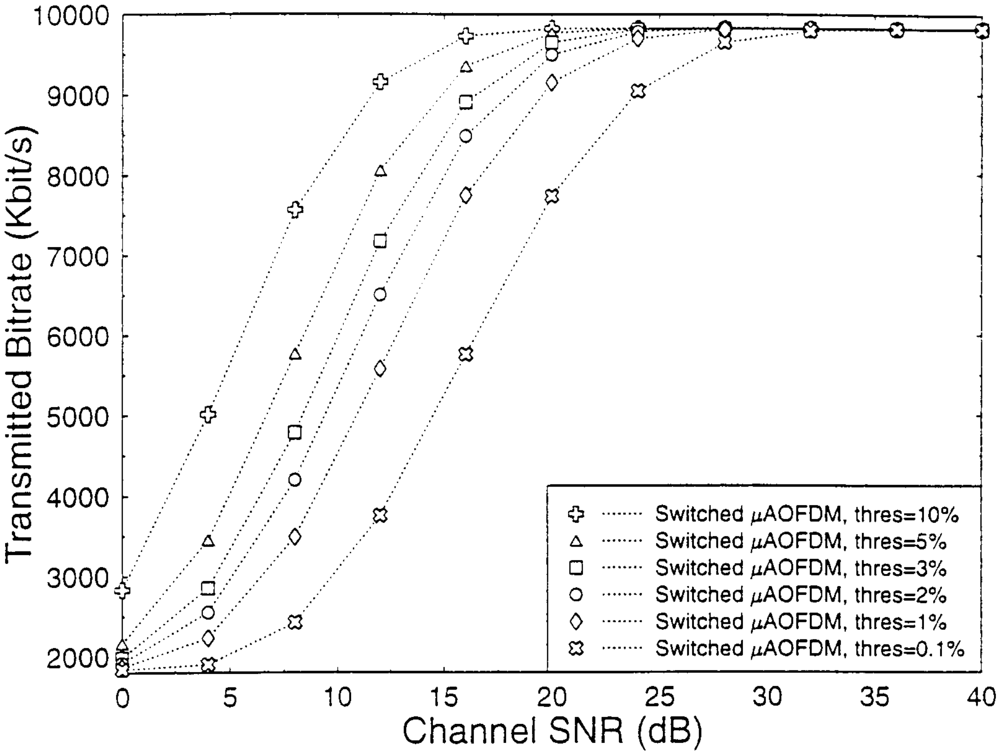

10000
9000
8000
7000
6000
5000
4000
3000
2000
Transmitted Bitrate (Kbit/s)
0
5
10
15
20
25
30
35
40
Channel SNR (dB)
Switched μAOFDM, thres=10%
Switched μAOFDM, thres=5%
Switched μAOFDM, thres=3%
Switched μAOFDM, thres=2%
Switched μAOFDM, thres=1%
Switched μAOFDM, thres=0.1%

Fig. 16

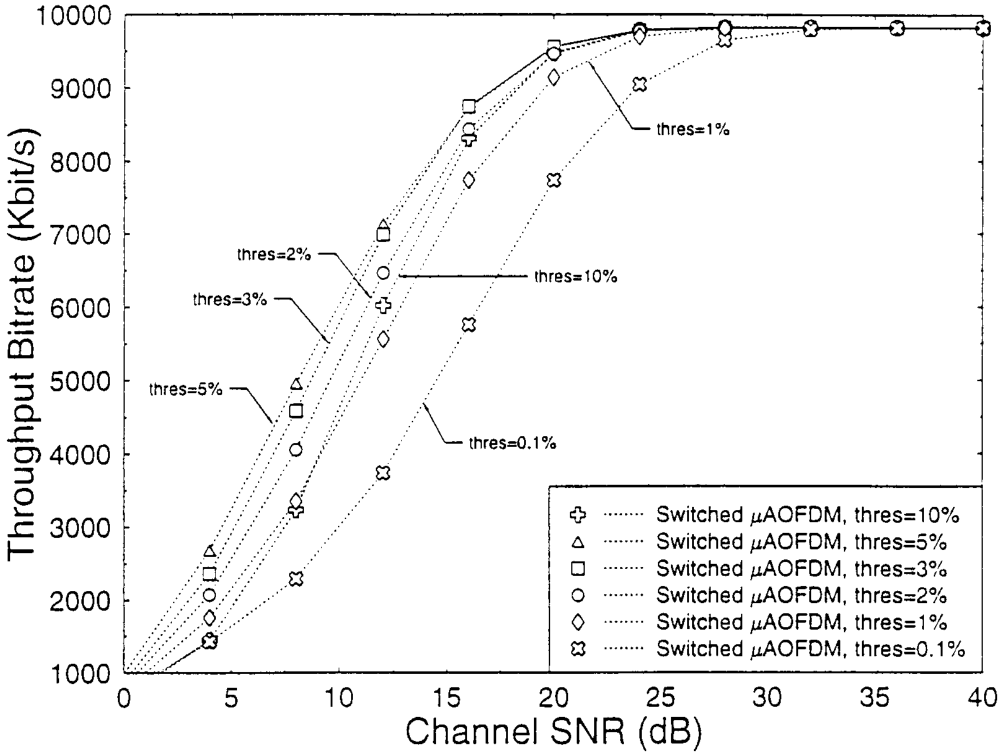

Throughput Bitrate (Kbit/s)
10000
9000
8000
7000
6000
5000
4000
3000
2000
1000
Channel SNR (dB)
0
5
10
15
20
25
30
35
40
thres=1%
thres=2%
thres=10%
thres=3%
thres=5%
thres=0.1%
✚ ········ Switched μAOFDM, thres=10%
△ ········ Switched μAOFDM, thres=5%
□ ········ Switched μAOFDM, thres=3%
○ ········ Switched μAOFDM, thres=2%
◇ ········ Switched μAOFDM, thres=1%
⌘ ········ Switched μAOFDM, thres=0.1%

Fig. 17(a)

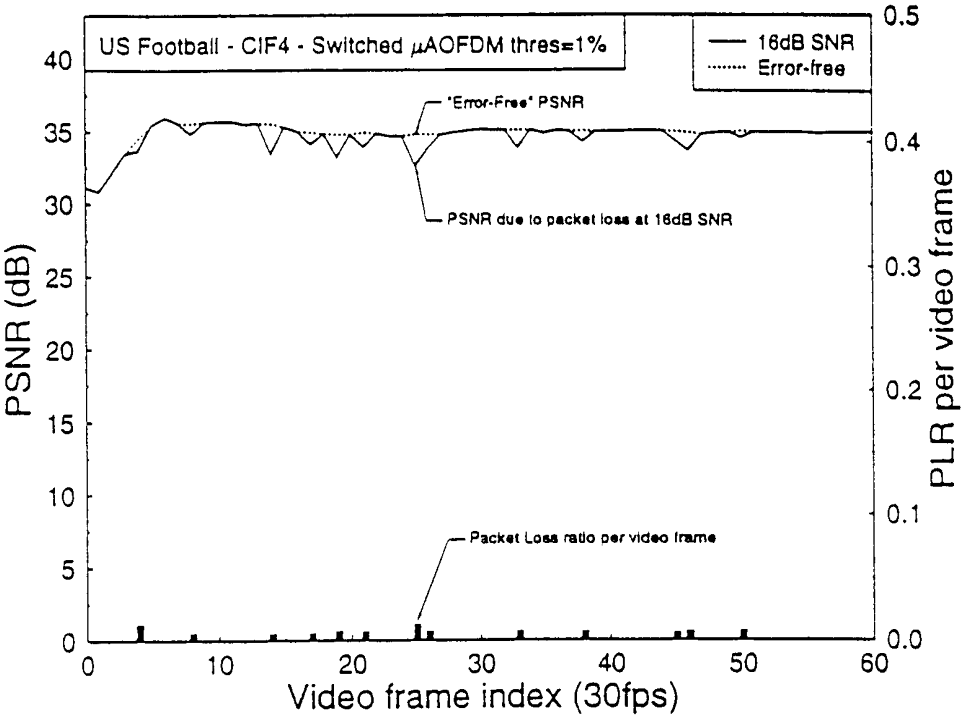


Fig. 17(b)

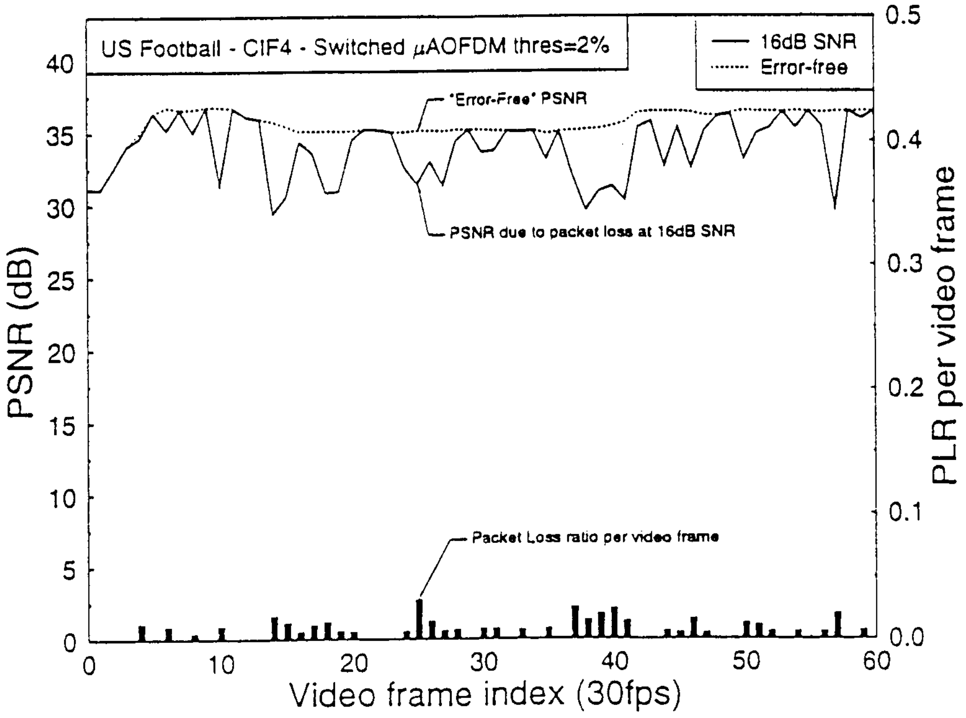


Fig. 17(c)

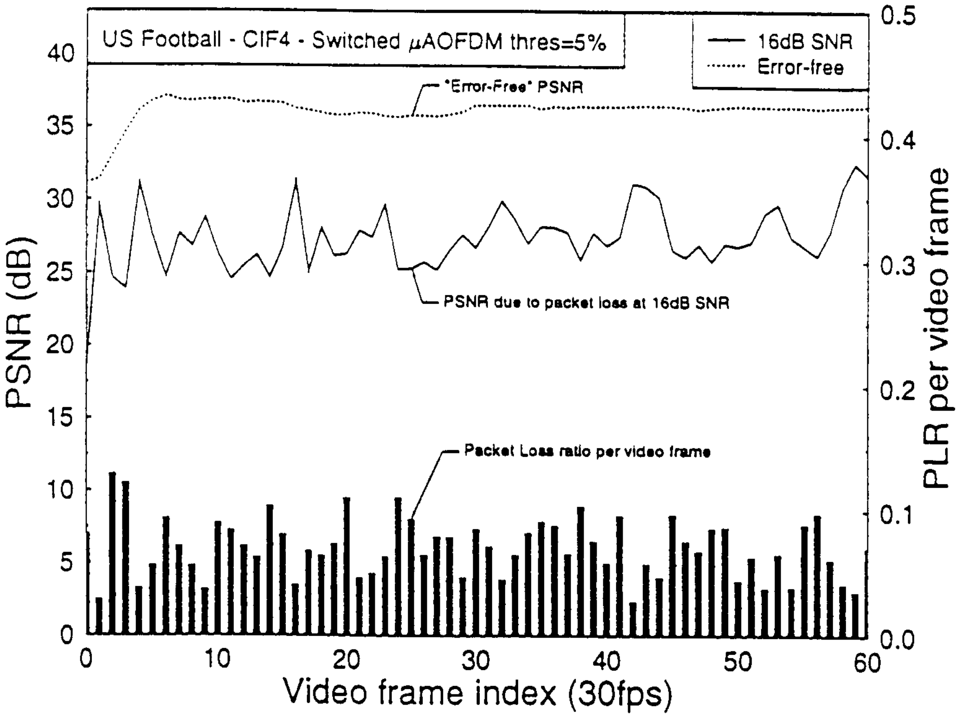

US Football - CIF4 - Switched μAOFDM thres=5%
16dB SNR
Error-free
"Error-Free" PSNR
PSNR due to packet loss at 16dB SNR
Packet Loss ratio per video frame
PSNR (dB)
PLR per video frame
Video frame index (30fps)
0
5
10
15
20
25
30
35
40
0.0
0.1
0.2
0.3
0.4
0.5
0
10
20
30
40
50
60

Fig. 18(a)

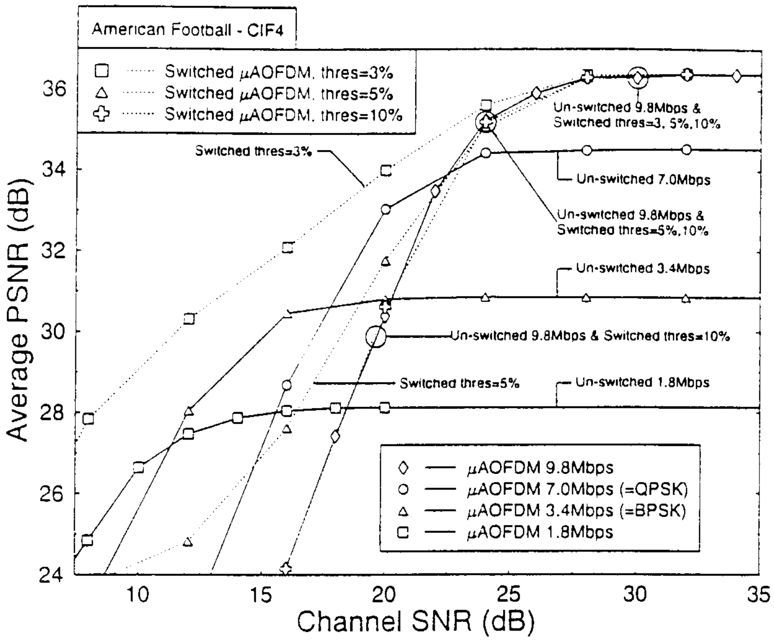
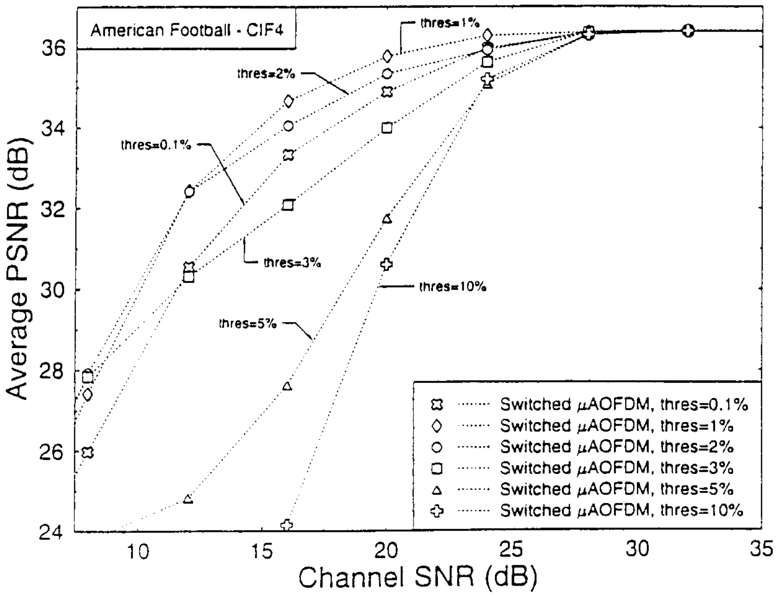


Fig. 18(b)



INTERNATIONAL SEARCH REPORT

Inte onal Application No
PCT/GB 00/01883

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04L27/26

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ, INSPEC, IBM-TDB, COMPENDEX

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	KELLER T ET AL: "Blind-detection assisted sub-band adaptive turbo-coded OFDM schemes" IEEE 49TH VEHICULAR TECHNOLOGY CONFERENCE, vol. 1, 16 - 20 May 1999, pages 489-493, XP002144718 Piscataway, NJ, USA, IEEE, USA ISBN: 0-7803-5565-2 the whole document ---	1-20
X	EP 0 869 647 A (LUCENT TECHNOLOGIES INC) 7 October 1998 (1998-10-07) abstract page 3, line 15 -page 4, line 2 page 4, line 17 - line 30 page 6, line 9 - line 48 page 7, line 2 - line 7 --- -/--	1-20

☒ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

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Date of the actual completion of the international search 11 August 2000	Date of mailing of the international search report 22/08/2000
Name and mailing address of the ISA European Patent Office, P.B. 5818 Patentlaan NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Authorized officer Koukourlis, S

1

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
X	US 5 479 447 A (CIOFFI JOHN M ET AL) 26 December 1995 (1995-12-26) column 11, line 10 -column 12, line 43	1-20
X	WO 99 16224 A (ERICSSON TELEFON AB L M) 1 April 1999 (1999-04-01) abstract page 1, line 19 - line 24 page 5, line 19 -page 6, line 2 page 7, line 22 -page 9, line 17 page 11, line 5 -page 17, line 4 page 19, line 20 -page 20, line 2 page 22, line 6 - line 13	1-9, 11-17, 19,20
P,X	KELLER T ET AL: "Sub-band adaptive pre-equalised OFDM transmission" IEEE VTS 50TH VEHICULAR TECHNOLOGY CONFERENCE, vol. 1, 19 - 22 September 1999, pages 334-338, XP002144719 Piscataway, NJ, USA ISBN: 0-7803-5435-4 page 334 page 337 - page 338; Section 2.5	1-20
P,X	L. HANZO, C.H. WONG, P. CHERRIMAN: "Burst-by-burst Adaptive Wideband Wireless Video Telephony" INTERNET ARTICLE, 'Online! 13 April 2000 (2000-04-13), XP002144720 Retrieved from the Internet: <URL:http://www-mobile.ecs.soton.ac.uk/hanzo/papers/invited-signal-proc-web.html> 'retrieved on 2000-08-10! page 7, right-hand column, paragraph 3 - paragraph 4 page 8, left-hand column, line 8 - line 24 page 8, right-hand column, paragraph 1 - paragraph 2 page 9, left-hand column, paragraph 1 -right-hand column, line 1; figure 6	1-20

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INTERNATIONAL SEARCH REPORT

Information on patent family members

Inte :onal Application No

PCT/GB 00/01883

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 0869647 A	07-10-1998	JP 10303849 A	13-11-1998
US 5479447 A	26-12-1995	NONE	
WO 9916224 A	01-04-1999	US 6005893 A	21-12-1999
		AU 9286698 A	12-04-1999
		EP 1018252 A	12-07-2000